



Prediction of room acoustical parameters (A)

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PROGRAM OF

The 121st Meeting of the Acoustical Society of America

Omni Inner Harbor Hotel • Baltimore, Maryland • 29 April–3 May 1991

MONDAY AFTERNOON, 29 APRIL 1991

LIBERTY B, 12:55 TO 5:00 P.M.

Session 10C

Acoustical Oceanography: Open Workshop on Gassy Seafloor Sediment: Field Measurements

Aubrey L. Anderson, Chair

Department of Oceanography, Texas A&M University, College Station, Texas 77843-3146

Chair's Introduction—12:55

Invited Papers

1:00

10C1. Interpretation of shallow gas-charged sediments on seismic records. Martin Hovland (Statoil, P.O. Box 300, N-4001 Stavanger, Norway)

Porewater and gas seepages through the seafloor have become the target of frontier research over the last decade. Such seepages occur worldwide and at water depths to over 5000 m. The seep locations are often identified by use of high-resolution shallow seismic systems. Data examples and case studies from the North Sea, the Skagerrak, offshore Mid Norway, and the Persian and Mexican Gulfs are presented and discussed. Such terms as "acoustic turbidity," "acoustic voids," "enhanced reflectors," and "wipe out zones" will also be discussed.

1:20

10C2. Anomalous acoustic behavior exhibited by gas-rich sediments of the subaqueous Mississippi Delta. Anthony R. Tinkle, Jeffrey A. May, Charles A. Meeder, and Kenneth R. Wener (Petroleum Technol. Ctr., Marathon Oil Co., P.O. Box 269, Littleton, CO 80160)

Sediments of the subaqueous Mississippi River Delta contain high concentrations of free gas, as manifested by anomalous physical and acoustic behavior. Much of the near-delta offshore area is a seismic "no-data zone," impenetrable to conventional profiling techniques. Specialized techniques developed for direct measurement of sediment acoustic properties reveal that velocities of less than 1000 ft/s (300 m/s) routinely occur in these muds, and acoustic energy at frequencies greater than 100 Hz is almost completely absorbed by only a few tens of feet of very gassy sediment. Severity of the acoustic anomalies typically decreases with depth beneath the seafloor, with the most severe anomalies generally occurring within the first 200 ft subbottom. Trapping of acoustic energy within these near-bottom, gas-charged layers is also observed. The nature and severity of the acoustic anomalies correlate well with both sediment physical properties, and the unstable system of collapse and mudflow features that typify the seafloor morphology of the deltaic no-data zone.

1:40

10C3. Field experiences in the direct measurement of anomalous acoustic behavior of gas-rich sediments. Anthony R. Tinkle, Charles A. Meeder, and Kenneth R. Wener (Petroleum Technol. Ctr., Marathon Oil Co., P.O. Box 269, Littleton, CO 80160)

Much of the seafloor in the immediate vicinity of the Mississippi Delta consists of sedimentary material containing high concentrations of free gas. Specialized techniques were developed for direct measurement of the acoustic properties of these gas-charged sediments, to sub-bottom depths of the order of 300 ft (100 m). These techniques included seafloor receiving arrays, modified checkshot profiles conducted in conventional engineering borings, and a seafloor-penetrating, hydrophone-instrumented rod used in the study of the shallowest sub-bottom layers. Collection of corresponding sediment samples was carefully integrated with these seismic experiments, to better determine the geologic nature and mechanical properties of the gas-charged materials. Final data acquisition practice was influenced both by the unusual physical properties of the materials being investigated, and the presence of strong and rapidly changing currents from the Mississippi River. Results of these *in situ* measurements demonstrate the widespread occurrence of anomalously low acoustic velocities (less than 1000 ft/s, or 300 m/s) and related extreme attenuation of higher frequency energy (greater than 100 Hz).

1:55

10C4. Effects of sediment gas on chirp sonar reflection profiles. Steven G. Schock, Lester R. LeBlanc (Dept. Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431), and Larry A. Mayer (Dalhousie Univ., Halifax, Nova Scotia B3H 4J1, Canada)

Chirp sonar reflection profiles of gassy and gas-free seabeds are quantitatively compared to show the effects of sediment gas on normal incidence backscattering measurements. Acoustic FM pulses that sweep over the band of 2 to 10 kHz are generated by the chirp sonar and compressed using a correlation filter to generate a bandlimited impulse response of the seabed. The amplitude spectrum of the ideal wavelet (the unattenuated, compressed FM pulse) and the spectrum of wavelets backscattered from gassy sediments are used to investigate the frequency dependence of backscattering from gassy seabeds. [Work supported by ONR.]

2:40-2:50

Break

Invited Papers

2:50

10C7. Near-surface gas hydrates in deep-sea sediments. Roger D. Flood (Marine Sci. Res. Ctr., SUNY, Stony Brook, NY 11794-5000), Patricia L. Manley (Middlebury College), and Mary I. Scranton (MSRC-SUNY, SB)

Recent studies of depositional processes in deep-sea continental margin and abyssal settings show that early diagenesis of organic-rich sediments can significantly alter physical properties in the upper 5–50 + m of the sediment column. Apparently, *in situ* methane production leads to sediment methane levels that may be high enough to allow hydrates to form while high CO₂ levels may cause localized carbonate precipitation.

2:10

10C5. Sound propagation in a shallow fresh-water aquaculture pond. Joe R. Zagar and Kenneth E. Gilbert (Natl. Ctr. for Physical Acoust., University, MS 38677)

Measurements made as early as 1943 have shown that gassy fresh-water sediments can be excellent reflectors of sound and behave acoustically as pressure-release surfaces. A simple propagation model is presented, based on bottom-reflection coefficients, that indicates one should expect strict mode propagation and sharp cutoffs in a horizontally stratified environment having gassy sediments. Continuous-wave measurements taken in a commercial aquaculture pond are presented and compared to the propagation model. The data show that, between 600 Hz and 3.0 kHz, a gassy sediment layer does behave as a nearly pressure-release surface. A simple mode-inversion technique is used to determine bottom-reflection coefficients for the sediments in the pond. Typical values for the bottom-reflection coefficient are found to be in the range of -0.85 to -0.95 . These values, as well as the estimates of the sediment sound speed, are in excellent agreement with those found by others. Additional measurements taken in depths as shallow as 7 cm are presented. These additional data suggest that the acoustic properties of the aquaculture pond that was studied are typical of all shallow water ponds possessing gassy sediments. [Work supported by ONR and USDA.]

2:25

10C6. Gas bubbles in ocean sediments and high-frequency acoustic backscattering strength. Nicholas P. Chotiros and Frank A. Boyle (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

The influence of trapped gas bubbles on the high-frequency acoustic bottom backscattering strength of ocean sediments will be examined. Statistical trends in the extant backscatter database at shallow grazing angles suggest that gas bubbles may be a dominant factor. The effect may be modeled in terms of an increase in the volume scattering strength of the sediment. Results from model and laboratory studies will be presented. [Work supported by ONR under NOARL management.]

These diagenetic processes can give rise to a series of near-surface reflecting horizons (and associated physical properties) that are not related to primary depositional processes. Where best studied (in the Argentine Basin at 5000- to 5500-m water depth [P. L. Manley and R. D. Flood, *Deep-Sea Res.* **36**, 611–623 (1989)]), sediment velocities as low as 1.35–1.42 km/s overlie velocities as high as 1.8–3.0 km/s within 30 m of the sediment surface. While the high velocities appear to be related to near-surface hydrate formation, the origin of the low velocities is less well understood. Similar reflecting sequences have been observed in other areas, particularly in the North Atlantic.

3:10

10C8. Seismic reflection velocity study of a gas-hydrate zone on the continental slope offshore South Carolina. Thomas H. Shipley (Inst. for Geophysics, Univ. of Texas, Austin, TX 78712), Warren Wood, and Paul L. Stoffa (Dept. of Geological Sciences)

The acoustical and physical significance of bottom-simulating seismic reflections (BSR's) remains an observational challenge to geophysical methods. A common depth point (CDP) seismic reflection profile using a 240-channel, 6000-m array with a 177-liter (5- to 60-Hz) source was collected along the continental rise off the eastern U.S. where a BSR reflection is observed along a small portion of the line at about 3200-m depth. These data provide some velocity estimates in the vicinity of the BSR. The CDP data were transformed to the domain of vertical delay time and horizontal ray parameter for velocity analysis purposes. Even so, the resulting velocity profiles have limited vertical resolution (about 200 m) due to the distribution of interpreted sedimentary reflections used in the vertical delay time velocity analysis. Even with this admittedly low vertical resolution, the velocity above the BSR is at least 2000 m/s in an approximately 200-m zone, while the predicted velocity based on the extrapolation of regional gradients indicates that normal sediments should have a velocity of about 1850 m/s. A velocity of 2000 m/s suggests on average about a 50% substitution of hydrate in the pore spaces but the actual vertical concentration gradient is not constrained. This velocity anomaly also extends into other areas just above the theoretical phase boundary position, but where there is no detectable BSR. Beneath the BSR, even with the relatively low vertical resolution, a velocity decrease to about 1700 m/s is detected. This low velocity is observed only in zones with a detectable BSR. It is not observed beneath the theoretical phase boundary position elsewhere. This suggests that the origin of the BSR is not a simple boundary between hydrated and nonhydrated, normal sediments below. Initial investigations of amplitudes indicate significant increase in amplitude with offset. Full waveform, offset modeling of the data is underway.

3:30

10C9. Estimation of amounts of gas hydrate in marine sediments using amplitude reduction of seismic reflections. William P. Dillon (U.S. Geological Survey, Woods Hole, MA 02543), Myung W. Lee (U.S. Geological Survey, Denver), Kristen Fehlhaber, and Deborah R. Hutchinson (U.S. Geological Survey, Woods Hole)

Gas hydrates (solid, crystalline water–gas mixtures) exist in sediments just below the seafloor. In seismic profiles, hydrate cementation creates zones of increased velocity and reduced amplitudes of stratal reflections (blanking). By using sediment velocities (estimated by an inversion method), known sediment porosity, and pure hydrate velocity, the amount of hydrate in the highest velocity and most intensively blanked sediments off the southeastern U.S. is calculated; this represents maximum hydrate cementation. To create seismic models of the range of possible blanking effects, ordinary, nonhydrated sediments across a reflecting boundary (caused by a porosity change) are “replaced” with this maximum-hydrate end member in various proportions. Three classes of blanking are defined; class boundaries represent a change in reflection amplitude by a factor of 2, and the classes are relatable to the amounts of hydrate in bulk sediment. In order to estimate the amount of hydrate, these classes are mapped in a grid of reflection profiles processed to preserve relative amplitude, and the first semiquantitative estimate of gas hydrates in deep-sea sediments is produced. [Work supported by DOE.]

3:50

10C10. Deep-towed acoustic measurements of gas hydrates. Mary M. Rowe and Joseph F. Gettrust (Naval Oceanographic and Atmospheric Res. Lab., Code 362, Stennis Space Center, MS 39529)

Acoustic data were collected using the Naval Oceanographic and Atmospheric Research Laboratory's (NOARL's) Deep Towed Acoustical/Geophysical System (DTAGS) at the northern end of the Blake Outer Ridge. A bottom-simulating reflector (BSR) is observed in these data at the same depth that it is found in surface-tow multichannel seismic sections from approximately 150 km to the east, near Deep Sea

Drilling Project (DSDP) sites 102, 104, and 533. Analysis of the high-resolution (~ 10 m) DTAGS data confirms that gas hydrate as well as unfrozen gas are concentrated within these sediments. Distinctive properties of the acoustic signal are combined with sediment sound speeds derived from these data to estimate the distribution and extent of the gas hydrate and regions of unfrozen gas. Results of analysis of DTAGS data extend findings from DSDP Leg 11, DSDP sites on the Middle America Trench, and the Black Sea. These data provide a means for determining mechanisms for the concentration of methane gas and the formation of gas hydrate. [Work supported by the Office of Naval Technology.]

4:05-4:15

Break

Poster Paper

The author will be at his poster from 2:40 to 2:50 p.m. and from 4:05 to 4:15 p.m.

10C11. Biogenic gas in shallow water sediments as indicated by high-resolution seismic records. Douglas N. Lambert (Naval Oceanographic and Atmospheric Res. Lab., Code 361, Stennis Space Center, MS 39529-5004)

A 15-kHz narrow-beam (12°) seismic system, designated the Acoustic Seafloor Classification System (ASCS), has been used to identify areas of shallow seafloor gassy sediments. The ASCS was designed to remotely classify seafloor sediments by quantitatively and qualitatively measuring the echo return amplitude and pulse character in ten adjustable time windows that correspond to depth increments in the sediment. In addition, the ASCS produces a very high-resolution analog seismic record of the upper few meters of the seafloor on which the amplitude of the echo return (echo strength) for each of the depth increments is

also indicated. This unique combination of data provides an excellent means of identifying sediments containing biogenic gas. Examples of identified gassy sediments shown on ASCS records range from a "speckled" appearance indicating random and isolated small gas concentrations to "white outs" that indicate large gas concentrations that completely attenuate the acoustic signal. Other records show highly reflective sediment layering that is due to a concentration of gas along bedding planes alternating with large pockets of low-reflectivity, degassed sediment that appears to have been "homogenized" by gas movement. These degassed sediment pockets are occasionally accompanied by vertical gas escape conduits seen in the sediment and by the presence of vertical reflectors in the water column indicative of escaping gas bubbles.

4:15-5:00

Panel Discussion

PANEL MODERATOR: Aubrey L. Anderson

PANEL MEMBERS: Martin Hovland
Anthony R. Tinkle
Roger D. Flood
Thomas M. Shipley
William P. Dillon

Session IID

Tutorial on Nonlinear Acoustics

Mauro Pierucci, Chair

*Department of Aerospace Engineering and Engineering Mechanics, San Diego State University,
San Diego, California 92182*

Chair's Introduction—7:00

Invited Paper

IID1. Nonlinear acoustics. Robert T. Beyer (Dept. of Physics, Brown Univ., Providence, RI 02912)

Nonlinear acoustics (NLA), within the framework of physical acoustics, has come of age over the past 30 years, yet new phenomena continue to be observed, so that the subject is not yet exhausted. The session will begin with a review of the mathematics of describing NLA (part 1). The analysis of one-dimensional phenomena will follow, plus the use of Burgers' equation (part 2). Data on the nonlinear parameter (B/A) will be reviewed (part 3). The physical aspects of nonlinear phenomena in two and three dimensions will be explored, and approximate methods of calculating the mutual effect of diffraction and nonlinearity will be described (part 4). The effects of the interaction of two sound beams will be considered (part 5). Applications will be treated. These include the development of parametric array sonar, surface wave devices, lithotripsy, and other medical applications (part 6). Such frontier topics as solitons and chaos will be touched on (part 7).

Session 2AA

Architectural Acoustics: Newer Measurement Procedures in Auditoria I

John S. Bradley, Chair

Institute for Research in Construction, National Research Council, Ottawa, Ontario K1A 0R6, Canada

Chair's Introduction—8:55

Invited Papers

9:00

2AA1. Ten years of newer auditorium acoustics measurements. J. S. Bradley and R. E. Halliwell (Inst. for Res. in Construction, Natl. Res. Council of Canada, Ottawa, Ontario K1A 0R6, Canada)

This paper reviews progress in making newer types of auditorium acoustics measurements over the past 10 years. All of the newer measures require impulse responses to be obtained and the merits and limitations of various measurement approaches will be discussed including the type of source, receiver, and processing that is used. Objective descriptions and the subjective relevance of the various types of newer measures will be reviewed including both monaural and binaural quantities. Four generations of measurement systems will be described including the special problems associated with measuring inter-aural cross correlation coefficients. Finally the questions of accuracy and repeatability will be mentioned along with some discussion of future developments and problems to be solved.

9:30

2AA2. Auditorium acoustics: What should we measure? What do we measure and what does it mean? J. P. Vian and X. Pelorsen (Room Acoustic Group, C.S.T.B. 24 rue J. Fourier, F-38400 St. Martin d'Hères, France)

For many years the assessment of auditorium acoustics has been made through a set of acoustical criteria whose number and kind sometimes differ substantially between groups of specialists. The situation seems quite clear for speech auditoria as much as it is confused for music auditoria. Meanwhile a growing consensus has slowly come out about a small number of quantities that are considered to explain a large amount of the variability of subjective judgments, because they correlate individually with some subjective impressions. The acoustical quality of an auditorium should be allowed to be expressed univocally by the values taken by these criteria. Unfortunately several of these criteria are not independent and very little is known about the subjective significance of the various arrangements of criteria values. Moreover all the variability observed when measuring the criteria are related not only to physical features of the auditorium but also to a large extent to the measurement technique. Different measurement techniques are discussed and fluctuations among measurement procedures, among halls and within halls are compared. It is shown how difficult it is to make a conclusion on acoustical quality of a hall by looking only at the criteria. The need for a standardized method to take measurements is stressed, as well as the need for more research work focusing on criteria, making it possible to discriminate between halls when listening judgments are different.

10:00

2AA3. Measurement of impulse response and its applications in room acoustics. H. Tachibana, H. Yano, and Y. Hidaka (Inst. of Industrial Sci., Univ. of Tokyo, Roppongi 7-22-1, Minato-ku, Tokyo 106, Japan)

Impulse response includes almost all physical information of a linear system and it is also very important in room acoustics. Fortunately, owing to the recent development of digital signal processing techniques and instrumentation, it has become possible to make a precise and convenient measurement of impulse responses in rooms. In this paper, the practical techniques for this kind of measurement including scale model studies are introduced. For the measurement in real auditoriums, a sweep pulse is radiated many times from a dodecahedral omnidirectional loudspeaker and the responses are recorded on a DAT through an omnidirectional microphone or a dummy head system. From the recorded responses, impulse responses are obtained by synchronous averaging and deconvolution techniques. In scale model experiments, impulse responses are measured by using a spark discharge source and a scale model dummy head microphone. From the impulse responses measured in such ways, not only various room acoustic quantities are derived but also the sounds including the room response can be synthesized by making convolutions between the

impulse responses and arbitrary dry source signals. This technique is effectively used for subjective evaluation of room acoustics. Some examples of the experimental results for real and scale model auditoriums will be demonstrated.

10:30

2AA4. Prediction of room acoustical parameters. A. C. Gade (Acoust. Lab., Danish Tech. Univ., Lyngby, Denmark DK 2800)

During a 6-yr period, detailed room acoustical measurements have been carried out in 35 halls in Denmark and in other European countries. By subjecting these data to statistical analyses, it has been possible to confirm old and establish new relationships between design variables and the position-averaged acoustical data. The results are presented in the form of linear, multiple regression formulas that may be used to predict the values of the newer measures of level, clarity, spaciousness, and musicians' conditions on the orchestra platform in halls with given RT and geometry.

11:00

2AA5. Subjective and objective evaluations of rooms for music. Richard P. Cervone, Wei-hwa Chiang, Gary W. Siebin (Dept. of Architecture, 231 ARCH, Univ. of Florida, Gainesville, FL 32611), Harold W. Doddington, and Wilhelm K. Schwab (Univ. of Florida, Gainesville, FL 32611)

A dedicated, computer-based analysis system was developed to perform a complete set of acoustical measurements of recent interest in full-size rooms and in scale models of rooms. The measurements included reverberation time, early reverberation time, loudness, early to late temporal energy ratios, lateral energy fractions, interaural cross correlation, and speech transmission index among others. Measurements were made at multiple locations in ten large concert halls. Groups of listeners evaluated live music performances at three locations in each of the rooms using a seven point semantic differential rating scale. Correlation analysis and statistical modeling identified significant relationships among the qualities of the music in the room rated by the listeners with the physical measurements made in the rooms. Variations of subjective qualities were identified among the different rooms and within each of the rooms as well. The subjective qualities that contributed to overall acoustical impression were also identified. [Work supported by NSF.]

11:30

2AA6. Subjective and objective assessment of some reverberation enhancement systems. Mendel Kleiner and Peter Svensson (Dept. of Appl. Acoust., Chalmers Univ. of Technol., S-41295 Gothenburg, Sweden)

How much of a difference does active or passive variation of a room's acoustical conditions make? Electronic architecture and reverberation enhancement have become common in the pursuit of better acoustics in multipurpose auditoria by active means. New systems show promising results. The properties of several different installations both in the U.S. and Europe have been investigated. This has been done both objectively by measurement of omnidirectional and binaural impulse responses and subjectively by the use of binaural recording. The binaural recordings used anechoic music replayed in stereo on stage. These recordings were used in pair comparison tests to investigate the dimensions of audible difference between the halls, with and without active or passive treatment of the room response. The pair comparison tests were evaluated using multidimensional scaling. The results show that some systems are able to modify the acoustical conditions to a very large extent. Differences between various system settings may be as large as between halls. Most of the subjectively perceived differences may be explained by changes in reverberation time and lateral energy factor. [Work supported by the Swedish National Council of Building Research.]

Session 2EA

Engineering Acoustics: Transducers and Arrays

James M. Powers, Cochair

Naval Underwater Systems Center, New London, Connecticut 06320-5594

George S. K. Wong, Cochair

Institute for National Measurement Standards, National Research Council, Ottawa, Ontario K1A 0R6, Canada

Chair's Introduction—8:45

Contributed Papers

8:50

2EA1. High-power test of a barrel stave flextensional transducer.

Mark B. Moffett and William L. Clay, Jr. (Naval Underwater Systems Ctr., New London, CT 06320)

Barrel stave flextensional transducers are potentially useful as compact, low-frequency, high-power projectors. An equivalent circuit model that includes a higher-mode, extensional compliance is used to estimate the maximum radiated power. Because the mechanical quality factor, Q , is low (on the order of 3 or 4), the source level of such a projector is limited by the maximum electric field that the piezoelectric ring stack driver can safely handle without depolarization or significant dielectric losses (about 400 kV/m for Navy type III lead zirconate titanate). A barrel stave flextensional projector 18 cm long and 9 cm in diameter with a mass of 4.1 kg in air was tested to 200 psig (1.4 MPa) in the pressure vessel at NUSC's Dodge Pond Field Station. A source level of 194.7 dB/1 μ Pa-m was obtained at 1.56 kHz for an applied rms voltage of 5 kV. The projector figure-of-merit was about 14 W/kg-kHz- Q , and this number would be expected to apply to a larger, lower-frequency projector of commensurate dimensions.

9:05

2EA2. Applications of very high-energy density electrostrictive ceramics for underwater projectors. A. P. Ritter, S. M. Pilgrim, P. Kuhn, S. R. Winzer (Martin Marietta Labs., 1450 S. Rolling Rd., Baltimore, MD 21227), and J. Sewell (Martin Marietta Aero & Naval Systems)

Performance of Navy sonar transducers is limited by the inherent energy density of the driver material—especially for those size-constrained applications that require very high source levels and/or very low frequencies. Ultimately, the maximum sound-pressure level will be limited by the amount of power that can be generated from the ceramic driver at its maximum engineering limit of 10–15 V/mil (~ 0.5 MV/m). Incremental improvements in transducer performance may be possible through design refinements, however, revolutionary large-scale improvements require new approaches to overcome the basic PZT material limitations. Electrostrictive ceramic materials, such as the PMN-based compositions being developed at Martin Marietta Laboratories, have energy density values an order of magnitude higher than Navy PZT's and therefore could significantly improve transducer performance (i.e., maximum attainable source level) if substituted for PZT in conventional transducer designs. Transducer model calculations for comparable PZT- and PMN-driven transducers show ~ 10 -dB gain in the transmitting voltage response for the PMN transducer. These model

predictions have been verified by experimental data obtained under a joint Martin Marietta/Navy materials development program.

9:20

2EA3. Using piezoelectric film and ultrasound resonance to measure the elastic moduli of spherical ceramic particles. P. S. Spoor, M. J. McKenna, and J. D. Maynard (Dept. of Physics, Penn State Univ., University Park, PA 16802), and John R. Hellmann (Penn State Univ., University Park, PA 16802)

The search for alternate sources of energy has prompted interest in small ceramic beads, called "proppants," which were developed as a means of "propping" open cracks during the hydraulic fracturing of bedrock in the vicinity of oil wells; recently, they have been considered as possible thermal transfer media for use in solar receivers [J. R. Hellmann *et al.*, "Evaluation of Spherical Ceramic Particles for Solar Thermal Transfer Media," SAND86-0981, Sandia National Laboratories, January 1987]. To monitor the effects of repeated thermal stresses on the proppants, one would like to have a reliable measure of their elastic constants; however, their spherical shape and small size (≈ 500 μ m) make conventional techniques, such as pulse-echo, inapplicable. Using a special piezoelectric film transducer and a small-sample resonance technique [J. D. Maynard, J. Acoust. Soc. Am. Suppl. 1 85, S20 (1989)], the authors have been able to determine the elastic constants and their variation as a function of heat treatment. [Work supported by the Office of Naval Research and NSF Grant DMR 9000549.]

9:35

2EA4. Optimally formulated high efficient planar projector arrays. P. M. Joseph and P. R. Saseendran Pillai (Dept. of Electron., Cochin Univ. of Sci. and Technol., Cochin 682 022, India)

Closely packed multi-element transducer arrays are extensively used in underwater applications for achieving better directionality and longer transmission range. With the current tendency of extending their operation toward lower frequency and particularly near resonance, the acoustic interaction among the elements grows stronger and in turn will degrade the predicted transmitting characteristics. This troublesome effect is much alleviated in uniform planar arrays by restructuring it with the optimal interelement spacing at which the interaction force is minimum [P. M. Joseph and P. R. Saseendran Pillai, Acoust. Lett. 12 (11), 190–193 (1989)]. It has been seen from the results of computation that a further reduction in interaction can be achieved by incorporating the nonuniform array concept. A simple method for predicting the opti-

mum configuration of element locations that reduce the interaction to a lower level, without altering the aperture dimension, is proposed in this paper. Here, the element locations are dispersed in accordance with Gaussian distributed random numbers by keeping the optimum spacing as the mean. Even though a large standard deviation value will greatly reduce interaction, a reasonable value is chosen, as large deviations produce deteriorated beams. This optimally formulated array does not exhibit any grating lobes nor does its radiation pattern differ much from the conventional $\lambda/2$ spaced and restructured arrays.

9:50

2EA5. The generation of sound by a dipole moving near the edge of a half-plane. Gerrit Schouten (Aerospace Dept., Delft Univ. of Technol., Kluyverweg 1, 2629 HS, The Netherlands)

The shape and the spatial amplitude distribution of the sound pulse generated by the passage of a dipole (or ring vortex) near the sharp edge of a half-plane are computed and presented in plots. Instead of using a low-frequency approximation to the Green's function a direct approach to the potential of a moving dipole in a branched space is used together with an image dipole. In a compressible medium (with a finite velocity of sound) the wave is physically present. In the limit of an incompressible medium the wavelength becomes infinite and the wave degenerates into a time history of the pressure variation. The incompressible pressures agree with those obtained from direct differentiations of the classical results of Sommerfeld [Proc. London Math. Soc. 30, 121-163 (1897)]. The wave results are in conflict with some results presented in the literature [T. Kambe *et al.*, J. Fluid Mech. 155, 77-103 (1985)] obtained with a low-frequency approach.

10:05

2EA6. Characterization of anisotropy in wood composites. V. Bucur (Centre de Recherches Forestières de Nancy, Champenoux, 54280 Seichamps, France)

The aim of this report is to study the anisotropic behavior of wood composites using the ultrasonic velocity method and acoustic emission method. Velocities of ultrasonic longitudinal and transversal waves were used for the estimation of five elastic constants. Complementary, stimulated acoustic emission, induced by breaking 0.5-mm pencil lead on the surface of the specimen, was employed to measure five parameters of the acoustic emission signal (duration, counts number, energy, amplitude, and rise time). The anisotropy was estimated as the ratio of velocities, of acoustic invariants, and acoustic emission parameters.

10:20

2EA7. Flow noise induced in large arrays via the flexure of the support. B. Dubus (Institut Supérieur d'Electronique du Nord, 41 Blvd. Vauban, 59046 Lille Cedex, France) and R. E. Montgomery (Naval Res. Lab., Orlando, FL 32856)

The noise induced by the turbulent boundary layer in sonar arrays is usually split into a direct path where the fluctuating pressures directly excite the hydrophones after traveling through an elastomer layer backed by a rigid surface, and the indirect path where the excitation of the support by the turbulent boundary layer induces the noise in the array. To describe this last phenomena, the force-modal transform method [M. C. Junger and D. Feit, *Sound, Structures, and Their Interaction* (MIT Press, Cambridge, MA)] has been extended to describe the flexure of a submerged multilayered plate and the wave-number spectrum of the noise sensed by the array. Several configurations are analyzed showing the effect of the stiffness and the damping of the support and of the distribution of the hydrophones. From these results, some solutions to reduce the noise level are proposed. [Work partially supported by Direction des Recherches Etudes et Techniques, Paris.]

10:35

2EA8. Wave propagation in a viscoelastic medium with a high concentration of pores. Y. Ma, V. K. Varadan, V. V. Varadan (Res. Ctr. for the Eng. of Electron. and Acoust. Mater. and Dept. of Eng. Sci. and Mech., Penn State Univ., University Park, PA 16802), and C. Audoly (G.E.R.D.S.M. Le Brusca, 83140 Six-Fours, France)

Wave propagation in a viscoelastic and porous medium, whose properties are temperature and frequency dependent, was investigated. The propagation characteristics of compressional waves in such a medium cannot be predicted using single scattering models when the porosity of the medium is considerable. Different from a previous study [Y. Ma, V. K. Varadan, and V. V. Varadan, ASME J. Heat Transfer 112, 402-407 (1990)], which was for electromagnetic waves, this investigation provided another proof of enhanced attenuation due to dependent scattering for the acoustic case. Most importantly, this is a problem of an elastic wave propagation in a lossy host medium with lossless scatterers which has seldom been considered. Numerical results of phase velocity as well as of attenuation of such a medium for different temperatures and different frequencies are presented. Differences in results between single and multiple scattering when the porosity is increased are also shown. Numerical results considering multiple scattering compare favorably with those from the experimental work which was done independently by G.E.R.D.S.M. in France. [Work supported by CEEAM.]

10:50

2EA9. Computation of sound radiation from an arbitrary body with mixed boundary conditions using program CHIEF. W. Thompson, Jr. (Penn State Univ., Appl. Res. Lab., P.O. Box 30, State College, PA 16804) and C. M. Siders (Naval Res. Lab., Orlando, FL 32856-8337)

The computer program CHIEF has been modified to allow the computation of sound radiation from an arbitrary-shaped body where the normal velocity is prescribed on only part of the surface of the body while a locally reacting complex impedance is assumed to cover the remainder of the surface. This impedance can range in value from zero (pressure-release boundary condition) to essentially infinity (rigid boundary condition). Different values of impedance are allowable on different portions of the same surface. Some results of computations of far-field directivity patterns and the radiation impedance loading on simple-shaped radiators with the mixed boundary conditions are presented. [Work sponsored by ONT.]

11:05

2EA10. Finite element model for a planar fiber optic hydrophone. Paul J. Zoccola and Paul C. Shang (Flow Noise Branch, David Taylor Res. Ctr., Bethesda, MD 20884)

A finite element model has been developed for estimating the acoustic sensitivity of a planar fiber optic hydrophone. The sensor consists of a single-mode fiber surrounded by a thin layer of silicon and a hytrel jacket. This strand is coiled and embedded in an elastomer layer. The model will allow use of nonisotropic materials also; these materials are less suitable for analytical modeling. The finite element model uses linear solid elements and uses symmetry about two or three planes, depending on the configuration. Because the sensor is designed for low-frequency measurements, hydrostatic pressure loading is used. Sensitivity is calculated using the strain on the fiber. The results are compared to closed form solutions for cylindrical configurations and to experimental results. They show that for most practical purposes, sensitivity is mostly controlled by the volume of the elastomer surrounding the fiber.

11:20

2EA11. Theoretical study of a fiberacoustic waveguide with continuous variation of the radial acoustic velocity. A. G. Yin, V. K.

In this paper, a geometrical acoustics method has been presented to describe acoustic wave propagation in a fiberacoustic waveguide in which the acoustic velocity varies continuously in the radial direction. Based on the acoustic wave equation, a WKBJ approximate method has been used to analyze the propagation and cutoff characteristics of guided acoustic modes in this waveguide. It is believed that this acoustic fiber can transmit acoustic and optical signals at the same time when the refractive index of the core material is greater than that of the cladding material and the acoustic velocity of the core material is less than that of the cladding material. Furthermore, this type of acoustic fiber is suitable because of low loss and hence long distance transmission paths.

11:35

2EA12. A unified reciprocity method for two basic types of vibration calibrators. Li-Feng Ge (Anhui Bureau of Standards and Metrology,

Hefei, Anhui 230001, People's Republic of China)

Two basic types of vibration calibrators, i.e., electrodynamic-type and piezoelectric-type standard vibration generators, can be described by a uniform dynamic model and equivalent circuit [L.-F. Ge, J. Acoust. Soc. Am. **86**, 210-214 (1989)]. The calibration factor of both types, which is defined as a ratio of the mechanical output (velocity or acceleration) to the electrical input (current or voltage), viz., the transmitting current or voltage response, is given and formulated by a uniform equation: $T_i = (CZ + D)^{-1}(\text{m s}^{-1}/\text{A})$ or $T_e = (AZ + B)^{-1}(\text{m s}^{-1}/\text{V})$, where Z is the mechanical load impedance and A , B , C , and D are the reciprocity network parameters of the calibrator. The former is suitable for an electrodynamic-type calibrator, because the thrust of the shaker depends on input current on the basis of the Faraday's law; the latter is suitable for a piezoelectric-type calibrator, since the voltage change leads to motion of the table according to the piezoelectric effect. Measurements performed on piezoelectric calibrators have obtained accurate calibration results at high-frequency range; experiments on a commercial electrodynamic shaker at 1 kHz have also shown that the method is attractive.

TUESDAY MORNING, 30 APRIL 1991

INTERNATIONAL E, 9:00 TO 11:20 A.M.

Session 2NS

Noise: Airport Noise Monitoring

Daniel L. Johnson, Chair

EG&G Special Projects, P.O. Box 9100, Albuquerque, New Mexico 87119

Chair's Introduction—9:00

Invited Papers

9:05

2NS1. The status of federal aviation noise regulations. John. E. Wesler (Wyle Labs., 2001 Jefferson Davis Hwy., Ste. 701, Arlington, VA 22202)

During its final hours, the 101st Congress enacted the Airport Noise and Capacity Act of 1990, mandating new controls on aircraft noise and airport access restrictions. The Act imposes new requirements for airport use restrictions that are imposed for noise control purposes, and establishes a schedule for the phase out of older, noisier, Stage 2 airplanes in the U.S. These legislative directives are now being translated into specific regulatory language by the Federal Aviation Administration. In summary, the actions continue the overall federal policies for aviation noise control that began in 1968. These policies are reviewed briefly, and the current status of the new regulations and restrictions is described.

9:30

2NS2. Airport noise monitoring for Boston area airports. Nancy S. Timmerman (Massport, 4th fl., Old Tower Bldg., Logan International Airport, Boston, MA 02128)

Airport noise has been monitored at Logan International Airport since 1975. Recently, Massport decided to install a new noise monitoring system. The new system will monitor trends in noise impact by time of day, season, and on an annual basis; measure noise levels generated by aircraft taxiing, ground operations, reverse thrust, takeoffs, landings, and overflights; obtain accurate data on aircraft flight tracks and fleet mix to be used in the generation of noise contours; and assess the performance of the Logan Preferential Runway Advisory System in terms of runway utilization and noise exposure. To do this, noise data from some sites will be obtained in 1/3-octave bands as well as in the traditional A-weighting; C-weighting is available; radar tracking information is expected to be part of the system; and weather data will be obtained at about half of the sites. All data are saved in database for study of the correlation of noise with various factors. A

complaint database, contour prediction mode (FAA integrated noise model), and mapping software are also part of the system. It is expected that this system will be used to evaluate different noise descriptors against actual complaints, and to suggest improvements to noise prediction models based on geography or meteorological effects.

9:55

2NS3. Airport noise and passive radar monitoring for Oakland International Airport. Glenn Woodman (Oakland International Airport, #1 Airport Dr., Box 45, Oakland, CA 94621)

Airport noise has been monitored at Oakland International Airport since early 1970s using mobile noise-monitoring terminals for one week sampling each calendar quarter in order to meet requirements of California state noise regulations. The Port of Oakland has recently installed a permanent airport noise and operations monitoring system that will enable achievement of three primary objectives: (1) gather acoustical data to meet California airport noise standards, (2) remove contribution of overflights from San Francisco International Airport to Oakland's noise exposure map, and (3) develop aircraft flight track measurements and positive identification of aircraft operators for community noise complaint resolution. Noise monitoring systems typically are passive in measuring only noise events. Oakland's system incorporates flight operations data and passive airport surveillance radar data to provide a proactive management tool for monitoring effectiveness of and compliance with existing noise abatement procedures. A discussion of this system will be made regarding its effectiveness in monitoring compliance with noise control procedures and in resolving community noise complaints.

Contributed Papers

10:20

2NS4. Intelligent aircraft noise monitoring terminals. Alan D. Wallis (Cirrus Res. Ltd., Hunmanby, England) and Robert Krug (Cirrus Res. Inc., Wauwatosa, WI 53213)

Current technology airport monitoring units using, "short L_{eq} " as the data transfer units, allow much more and better data than has been previously possible. With very large internal stores, in the order of 2 megabytes, the new generation of units can simultaneously store raw data elements and full environmental information as well as each aircraft event. The pattern recognition built into the units will allow each noise monitoring terminal on the system to recognize aircraft flying over. Then, by a template comparison technique, it both stores the resultant data and transfers it to a remote host for correlation. The terminal incorporates some learning capability to increase the "hit rate" of recognition. Any, or all, of the acquisition parameters in each terminal can be configured from the host using a modem link. In a similar way, diagnostics can be performed from the host without visiting each terminal. Typically, 1 to 2 weeks of raw data and up to 10 000 aircraft events are stored in each terminal. This allows for unattended operations as well as security backup for the host.

10:35

2NS5. Community aircraft noise assessment to determine structural noise attenuation. David Draper (Dresdner, Robin & Associates, Inc., P.O. Box 469, Jersey City, NJ 07302) and Henry Young (Young Environmental Sciences)

A community aircraft noise assessment was conducted for the City of New York for a major waterfront residential development (7000–8000 units) proximate to Kennedy International Airport in Queens. The development site lay beneath one flight path and immediately to the west of a second. Characterization of the site for aircraft noise by monitoring alone was difficult since the overflying runway was typically used during inclement weather during which monitoring conditions were at their worst. Thus, a combination of monitoring and modeling was conducted to determine the noise exposure of proposed structures and open space. Aircraft noise was monitored at three locations on the site. Modeling of a worst case scenario was performed using the aircraft noise database from the integrated noise model (INM). The model was "cal-

ibrated" by simulating specific aircraft flights, predicting to monitored locations and comparing the predicted and monitored noise levels. These predicted levels were used to determine the degree of structural noise attenuation required to meet New York City's interior residential noise requirements of 45 dBA.

10:50

2NS6. The effect of onset rate on aircraft noise annoyance. Kenneth J. Plotkin, Kevin A. Bradley (Wyle Labs., 2001 Jefferson Davis Hwy., Ste. 701, Arlington, VA 22202), and John A. Molino (Tech-U-Fit Corp., Alexandria, VA 22314)

Noise from military aircraft operating at high speeds and low altitude can exhibit high onset rates, which are thought to increase the annoyance of these sounds. A set of listening experiments was undertaken to examine this effect and to evaluate the validity of an onset rate adjustment currently used in the environmental assessment of this type of noise. A basic set of 12 stereo sound recordings was prepared, consisting of four types of military aircraft with various onset rates plus one civil aircraft. These aircraft flyby sounds were presented at four sound levels to subjects in an indoor listening facility (nominal SEL of 95, 85, 75, and 65 dB) and at an outdoor facility (nominal SEL of 115, 105, 95, and 85 dB). Indoor sounds were filtered according to a typical residential noise reduction curve. Sounds were presented in random order, at random time intervals, and random approach from either in front of or behind the subjects. Subjects rated each sound on a seven-point annoyance scale. Two companion experiments were also performed at the outdoor facility. These experiments used modified military aircraft sounds with particular onset rates from 5 to 100 dB/s and decay rates from 2 to 30 dB/s. Analysis of the results of these experiments examined the effects of onset and decay rates, level, SEL, and duration. [Work sponsored by USAF AAMRL/BBE.]

11:05

2NS7. An exploratory study of community noise levels in the city of Lima. Carlos R. Jiménez-Dianderas and Iván F. Rivas-Tejeda (Garcilaso de la Vega 163, Salamanca de Monterrico, Lima 3, Perú)

Lima, the capital city of Perú, a metropolis with more than 6 million

people, has in the last 30 years had a sudden, explosive, and unsettled growth. Consequently, the city is always attacked by an endless number of noise sources. The absence of requirements for acoustic planning has caused noise producing activities to be mixed in with buildings requiring a quiet sonic environment. This study analyzes the main community noise sources and their levels: traffic and transportation noise, aircraft, and industrial noise. The noise pollution levels obtained, which were calculated through theoretical formulas and empirical methods, are

plotted in noise maps of the metropolitan area of Lima. Unfortunately, Perú does not have an acoustical research center, laboratories in acoustics, or professional researchers in acoustics. The results of this research allowed comparison of the average levels calculated, around 92 dB in the downtown area, in regard to requested noise levels. The lack of legislation, acoustic research, and people's knowledge about noise damage will lead to its progressive increase, almost 1 dB per year, in the near future.

TUESDAY MORNING, 30 APRIL 1991

CARROLL, 8:30 TO 11:30 A.M.

Session 2PA

Physical Acoustics: Bubbles and Drops

Charles C. Church, Chair

National Center for Physical Acoustics, Coliseum Drive, University, Mississippi 38677

Contributed Papers

8:30

2PA1. Study of surface dynamics of drops in the presence of surfactants. Yuren Tian, Glynn Holt, and Robert E. Apfel (Dept. of Mech. Eng., Yale Univ., 2159 Yale Station, New Haven, CT 06520)

The characteristics of capillary waves, especially energy dissipation, are strongly influenced by the presence of surfactants. The phenomenon has been studied by investigating experimentally the free quadrupole oscillations of a fluid drop acoustically levitated in the air. The experimental apparatus is similar to that of Trinh [Rev. Sci. Instrum. **56**, 2059 (1985)], and the methods follow those reported by Lu and Apfel [J. Colloid Interface Sci. **134**(1), 245 (1990)]. Using this system, the frequency of free quadrupole oscillations and its damping constant can be measured. Water drops with different sizes and different surfactant concentrations were used in the measurements. Experiment results show that the technique used here, which is nonperturbative and requires a very small amount of sample, may supplement other methods to measure the surface properties of liquid. [Work supported by NASA through JPL, Contract 958722.]

8:45

2PA2. Stimulation of drop coalescence with acoustic cavitation. Edward A. Gardner and Robert E. Apfel (Dept. of Mech. Eng., Yale Univ., Box 2159 Yale Station, New Haven, CT 06520)

A standing acoustic field was used to levitate a pair of anisole (methoxybenzene) drops in an aqueous surfactant solution containing Teflon particles to act as cavitation nuclei. The drops were brought into contact with each other by the levitation field but did not coalesce immediately because of the surfactant in the water. By increasing the intensity of the sound field, cavitation events were produced, some of which ruptured the oil/water interface, causing plumes of micron-sized oil droplets to be expelled from one of the main drops. Occasionally, this disturbance caused the main drops to coalesce. This method of stimulating coalescence was studied using high-speed film and statistical analysis, which will be reviewed in the presentation. [Work supported by NASA through JPL, Contract 958722.]

9:00

2PA3. Collapse of a long cylindrical bubble. Jorge E. Lopez de Cardenas (Schlumberger Perforating Ctr., 10910 Airline Rd., Rosharon, TX 77583) and Robert D. Finch (Univ. of Houston, Houston, TX 77204)

The collapse of a cylindrical bubble was modeled using an approach similar to Rayleigh's solution for a spherical bubble. Rayleigh assumes an incompressible fluid, which, in the cylindrical case, leads to the problem of the fluid having infinite kinetic energy. To overcome the difficulty a "semi-compressible" assumption is used, in which only the fluid contained in the acoustic envelope defined by $r = ct$ is considered in the solution; the fluid outside this envelope being neglected. The results obtained with this approach were compared with solutions computed from a model using a finite element method. As in Rayleigh's solution for the spherical bubble, the calculations of the collapse time for the cylindrical bubble showed very good agreement with the numerical solutions. The results obtained for the pressure generated by the collapse of the cavity provide a qualitative description of cavitation effects produced by jets from shaped charges.

9:15

2PA4. A comparison between "real" and "ideal" gas in theoretical cavitation dynamics. Charles C. Church (Natl. Ctr. for Physical Acoust., Coliseum Dr., Univ. of Mississippi, University, MS 38677)

Most theoretical formulations for the response of small gas bubbles to acoustic pressure fields assume that the ideal gas equation of state is appropriate for calculating the internal pressure of the bubble. While this assumption is adequate at low amplitudes, at higher pressure amplitudes, and thus larger bubble responses, it leads to predictions of internal gas densities that are on the order of, or greater than, those of metals. A more realistic assumption is a van der Waals equation of state for a "real" gas. In the present work a general expression for the pressure inside a bubble containing real gas is provided, as well as expressions resulting from some common simplifying assumptions. In addition, comparisons between calculated bubble responses using either an ideal gas or van der Waals equation are presented. For these computa-

tions one of two models for bubble dynamics will be employed: the Gilmore-Akulichev formulation with temperature estimated polytropically or Flynn's equations in which temperature is calculated exactly. [Work supported by NIH and ONR.]

9:30

2PA5. Nonlinear pulsations of cavitation vapor bubbles near resonance. R. Edward Nicholas and Robert D. Finch (Dept. of Mech. Eng., Univ. of Houston, Houston, TX 77204-4792)

Studies of the nonlinear pulsations (and sometimes collapse) of cavitation bubbles near resonant size in an ultrasonic field were performed through a numerical solution of nonlinear equations describing the motion of a spherical vapor bubble. A number of plots are presented showing bubble pulsations. These plots show pulsations at the driving frequency with rich harmonic content but also show nonchaotic pulsations at frequencies below the driving frequency as the bubble grows through resonant size. The frequencies observed below the driving frequency seem unrelated to the driving frequency and instead vary with the amplitude of the ultrasonic field.

9:45

2PA6. Signal processing to extract transient microcavitation signals from noise. Qihong Xu, Christy K. Holland, and Robert E. Apfel (Dept. of Mech. Eng., Yale Univ., New Haven, CT 06520)

The time history of the scattered signal from transient microcavitation is different from the signal scattered by other targets, which constitutes most of the noise. The backscattered signals from transient microcavitation (bubbles of micron scale that last on the order of microseconds) are both frequency modulated and amplitude modulated; therefore, a Doppler technique can be one alternative to the detection of transient cavitation, which takes advantage of the fact that the signal is frequency modulated. Numerical simulations of these processing techniques will be presented. [Work supported by NIH Grant 5ROICA39374.]

10:00

2PA7. An experimental test of a theory of nonlinear bubble dynamics. Darren L. Hitt, Andrea Prosperetti (Dept. of Mech. Eng., Johns Hopkins Univ., Baltimore, MD 21218), and Ronald A. Roy (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

Nonlinear bubble oscillations in a levitation cell are studied. The experimental levitation number, $Le = \rho g / |\nabla p|$, can be compared to its corresponding theoretical value to provide a sensitive evaluation of the bubble dynamics theory [Prosperetti *et al.*, J. Acoust. Soc. Am. **83**, 502-514 (1988)]. A pulse-echo technique is developed that permits extremely accurate measurements of the bubble location in the pressure field. This technique can then be used to obtain accurate measurements of the levitation number under a variety of experimental conditions. [Work supported by NSF.]

10:15

2PA8. Rectified diffusion during nonlinear gas bubble oscillations. Vinod Kamath and Andrea Prosperetti (Dept. of Mech. Eng., Johns Hopkins Univ., Baltimore, MD 21218)

Mass transfer of the dissolved gas across a bubble wall during nonlinear spherically symmetric oscillations is studied. Use of an earlier comprehensive model [Prosperetti *et al.*, J. Acoust. Soc. Am. **83**, 502-514 (1988)] to determine the bubble dynamics avoids the need to use *ad hoc* polytropic assumptions. A numerical technique based on a pseudospectral method is used to determine the dissolved gas concentration

in the liquid. This formulation does not require the assumptions made in the classic theory of Eller and Flynn [J. Acoust. Soc. Am. **37**, 493-503 (1965)]. In some conditions results indicate a significant difference in growth rate predictions when compared to the Eller-Flynn model. [Work supported by NSF.]

10:30

2PA9. Nonlinear oscillations of gas bubbles in elastic solids. Paul B. Massell (Automation Counselors, Inc., 650 Americana Dr., Annapolis, MD 21403) and Gary Albin (Naval Med. Res. Inst., Bethesda, MD 20889-5055)

This work was part of a project whose long-range goal is the development of a device for estimating the size of bubbles in human tissue. Calibration standards are expected to consist of gas bubbles in polymer gels. Accordingly, equations in Prosperetti [J. Acoust. Soc. Am. **56**, 878-885 (1974)] that describe forced oscillations of a gas bubble in a liquid have been modified to apply to the case of a gas bubble in a linearly elastic solid. The changes in the analytic expressions for the resonance solutions are modest, and the major features are preserved (e.g., hysteresis for main, subharmonic, and harmonic frequency regions). More precise equations for the liquid case in Kamath and Prosperetti [J. Acoust. Soc. Am. **85**, 1538-1548 (1989)] have also been modified to treat the elastic solid case. Following these authors, a Galerkin spectral method was used to generate numerical solutions to the governing partial differential equations. Situations in which two or more stable solutions exist for a given set of parameters may be discussed.

10:45

2PA10. Experimental studies of the enhanced effective nonlinearity parameter B/A of a bubbly medium. Junru Wu and Zheming Zhu (Dept. of Physics, Univ. of Vermont, Burlington, VT 05405)

It has been known for some years that the presence of bubbles may enhance the nonlinearity parameter B/A of a medium. Two distinct physical mechanisms are believed to be involved. The first one is due to bubble nonlinear oscillations; bubbles are driven by an acoustic wave near resonance frequency. The second one is due to static properties of a bubbly medium. The effective nonlinearity parameter B/A of water containing a three-dimensional ensemble of randomly distributed uniformly-sized trapped cylindrical bubbles was measured. The measured effective B/A for the system is of the magnitude of 10^4 to 10^5 . The experimental results also suggest that the dramatic enhancement of the effective nonlinear parameter B/A is mainly due to the nonlinear resonance oscillation of the trapped bubbles. [Work supported by the NIH via Grant No. CA42947 and by the NSF and Vermont Epscor.]

11:00

2PA11. Calculations of temperature within and without a pulsating gas-filled cavity. Charles C. Church (Natl. Ctr. for Physical Acoust., Coliseum Dr., Univ. of Mississippi, University, MS 38677)

The temperature of the gas within a spherical cavity driven by a sinusoidal pressure field is calculated using an existing model [H. G. Flynn, J. Acoust. Soc. Am. **57**, 1379-1396 (1975)] for cavitation dynamics. The temperature field in the liquid surrounding the bubble also is calculated with this model. For this work it is assumed that temperature variations are due to heat conduction only, although corrections for evaporation, condensation, etc., may be included if desired. Because the temperature T in the liquid at the cavity interface is allowed to vary (which reduces heat conduction by reducing the thermal gradient at the interface) the effect of holding T fixed is investigated. Results are presented for argon-filled and air-filled cavities in water or various organic liquids. [Work supported by ONR and NIH.]

2PA12. Catalase and cysteamine inhibit indirect sonochemical effects of cavitation on cellular DNA. D. L. Miller, R. M. Thomas, and M. E. Frazier (Biol. and Chem. Dept., Battelle PNL, P.O. Box 999, Richland, WA 99352)

Ultrasonic cavitation can indirectly cause cellular bioeffects due to the production of toxic sonochemicals. Phosphate buffered saline (PBS) was exposed to 1.61-MHz ultrasonic cavitation at 20 °C in a rotating-tube exposure system to build up sonochemical products. Single strand DNA breaks (SSBs) were then induced by treating Chinese hamster ovary (CHO) cells with the cavitated PBS for 30 min on ice. This

indirect effect could be explained by the action of cavitation-generated hydrogen peroxide (e.g., up to 16 μM after 30-min exposure) in the PBS. Adding catalase to the cavitated PBS before cell treatment eliminated the effect. Tests with hydrogen peroxide showed that 16 μM H_2O_2 treatment for 30 min on ice was as effective as 1 Gy of gamma rays in producing SSBs. The SSB effect of H_2O_2 was reduced by addition of the radical scavenger cysteamine to the cells before treatment. These results indicate that DNA effects from ultrasonic cavitation can occur indirectly without the highly destructive direct interaction of cells and cavities. [Supported by PHS Grant CA42947 awarded by the National Institutes of Health.]

TUESDAY MORNING, 30 APRIL 1991

INTERNATIONAL C, 9:00 TO 11:30 A.M.

Session 2PP

Psychological and Physiological Acoustics: Models and Mechanisms of the Auditory Periphery

Bradford D. May, Chair

Department of Otolaryngology—HNS, Johns Hopkins University School of Medicine, Baltimore, Maryland 21204

Contributed Papers

9:00

2PP1. Sound intensity and reflection coefficient measurements in the ear canal. R. D. Rabbitt and J. Dragicevic (Depart. of Mech. Eng., Washington Univ., St. Louis, MO)

The sound power per unit cross-sectional area (specific power) is measured in the ear canal using a three-probe-microphone system. Pressure measurements recorded from locations within the core region of the ear canal are utilized to determine the instantaneous intensity of sound waves traveling in the lengthwise direction. A direct time-domain method provides the specific power contained in waves traveling through the region of the canal containing the microphones. The time average of the instantaneous intensity defines the specific power absorbed by the ear. Furthermore, the time average of the positive part of the intensity determines the incident specific power, or input to the ear, and the time average of the negative part of the intensity determines the reflected, or emitted, specific power. Signal averaged data are collected for pure-tone sweeps from 4 to 15 kHz. [Work supported, in part, by the Whitaker Foundation and the National Science Foundation.]

9:15

2PP2. Pressure transfer function from the diffuse field to the human infant ear canal. Douglas H. Keefe, Edward M. Burns, Jay C. Bulen, and Shari L. Campbell (School of Music, DN-10, and Dept. of Speech and Hearing Sci., JG-15, Univ. of Washington, Seattle, WA 98195)

The diffuse field pressure transfer function $H(f)$ to the ear canal of human infants was measured from 100–11 000 Hz. The diffuse field pressure and the ear canal pressure were simultaneously measured in a reverberant room using spatial averaging of source and microphones.

The probe microphone tip was inserted 5 mm into the infant ear canal, and the parent was instructed to hold the infant while slowly walking around the room; an experimenter walked around the room with the other microphone. Results using KEMAR and in adult canals were consistent with the literature. $H(f)$ for the 1-month-old infants has a strong peak (10–15 dB) at 4–5 kHz. The majority of the ten infants showed a twin peak structure, with the additional peak (8–12 dB) at 5–6 kHz. $H(f)$ has a dip at 8–9 kHz and increases up to 11 kHz. A 5-dB peak at 1–2 kHz may be due to the proximity of the parent's torso to the infant's head. Additional results on older infants will be discussed. [Work supported by NIH.]

9:30

2PP3. A physical model for the middle ear cavity (MEC). Sunil Puria (AT&T Bell Labs, 600 Mountain Ave., Murray Hill, NJ 07974 and The City College of CUNY, New York, NY 10036)

The input impedance of the cat MEC is modeled by approximating the bulla and tympanic cavities as a cascade of two cylindrical tubes. The small hole in the bony septum, known as the foramen, is approximated as a short cylindrical tube. Nonplanar wave propagation, due to the foramen, and visco-thermal losses at the cavity walls are included in the model. The model was verified by making impedance measurements on cylindrical cavities for frequencies up to 12 kHz. The intact case, as well as the open bulla case and plugged foramen case, is simulated and shown to be in good agreement with animal measurements [T. J. Lynch, III, Ph.D. thesis, MIT (1981)]. In addition to the bulla resonance near 4.5 kHz, the model predicts that there is a second bulla resonance near 26 kHz. Without the bony septum, the resonances are shown to be near 10 and 20 kHz. It is hypothesized that the role of the foramen is to shift the resonant frequencies.

2PP4. Middle ear muscle effects during gunfire noise exposures. G. Richard Price (U.S. Army Human Eng. Lab., Aberdeen Proving Ground, MD 21005-5001)

A previous study using evoked response audiometry in anesthetized cats [Price and Wansack, *J. Acoust. Soc. Am.* **86**, 2185 (1989)] showed that an impulse with its peak energy in the 4.0-kHz region was surprisingly hazardous. The surprisingly great effect could have been due to the anesthesia itself or the anesthesia could have prevented middle ear muscle contractions (normally not thought to affect the ear's response to gunfire). In order to help discriminate, three groups of ten cats were exposed to 50 midrange impulses at 145 dB peak, fired 3–5 s apart. Group I was anesthetized at the time of the exposure, and groups II and III were not; however, group II was anesthetized for threshold testing immediately after exposure (measured by evoked response audiometry at 2.0, 4.0, 8.0, and 16.0 kHz). Mean threshold shifts at 4.0 kHz 20–30 min post-exposure were 72 dB for group I and 19 dB for group II. Permanent threshold shifts were 44 dB for group I and 5.9 and 5.3 dB for groups II and III, respectively. These data suggest that the anesthetic itself probably does not potentiate the exposure and that the middle ear muscles should be suspected as being protective in any waking cat.

10:30

2PP6. The effect of salicylate on cochlear outer-hair cell osmoregulation. Mark E. Chertoff and William E. Brownell (The Ctr. for Hearing Sci., The Johns Hopkins Univ. School of Medicine, Baltimore, MD 21205-2195)

In vitro experiments have demonstrated that cochlear outer-hair cells (OHCs) can detect an increase in the extracellular osmotic pressure and, in response, regulate their internal pressure. Salicylate exposure reduces the turgor in electrically stimulated OHCs suggesting an interference with turgor regulation. Cytoplasmic pressure was determined by placing OHCs in a phosphate-buffered saline (PBS) solution and increasing the osmotic pressure by dripping a hyperosmotic solution (PBS plus 130 mM raffinose; Osmol = 410–440 mmol/kg) into the chamber. The raffinose perfusion was stopped when cell volume decreased and the osmolarity of the chamber was measured with a vapor pressure osmometer. The cytoplasmic pressure of OHCs in the PBS was 16.1 mmol/kg (± 5.7) greater than the PBS. When OHCs were exposed to a PBS solution containing an extra 10 mM NaCl or either 5 or 10 mM Na-salicylate, the pressure difference between the cytoplasm and the bathing medium was decreased to between –4 to 6 mmol/kg. After 10 min, the pressure difference increased to 11.3 mmol/kg (± 5.2) for the NaCl solution and 10.3 mmol/kg (± 4.16) for the salicylate solution. [Supported by ONR and NIDCD.]

10:45

2PP7. Frequency differences in the development of protection against NIHL by low-level "toughening" exposures. Malini Subramaniam, Donald Henderson, and Vlasta Spongr (Hearing Res. Lab., Dept. of Commun. Disord. and Sci., 215 Parker Hall, State Univ. of New York, Buffalo, NY 14214)

The aim of this study was to determine if low-level intermittent

2PP5. A new cochlear model: Viscous fluid motion. Wanggen Wan (c/o Yu Guoan, Xian Petroleum Inst., Xian, Shaanxi 710061, People's Republic of China) and Changxin Fan (Xidian Univ., Xian, Shaanxi 710071, People's Republic of China)

The cochlear fluid is considered to be actual viscous fluid in this paper and the Navier–Stokes equation used to describe the viscous fluid motion is linearized and then transformed to a typical transport equation and a Poisson equation. By solving the two equations using the Fourier series method, a new two-dimensional cochlear model is finally derived. The frequency responses of the model are presented and believed to be much more improved than those derived before, especially in the low-frequency slope. Therefore, they are more consistent with recent experimental data [Robles *et al.*, *J. Acoust. Soc. Am.* **80**, 1364 (1986)]. It appears that the inclusion of the fluid viscosity sharpens the low-frequency slope of the cochlear model. The high-frequency slope is not as sharp as desired for the effects of the middle ear and the basilar membrane (BM) nonlinearity (signal-dependent damping) are not included. The new cochlear model also makes such inclusion feasible.

10:15–10:30

Break

exposures to a 0.5-kHz OBN has a protective effect for future high-level exposures (1) at the same frequency or (2) at a higher frequency (4-kHz OBN). Four groups of monaural chinchillas were used as subjects. Thresholds were determined using evoked potentials recorded from a chronic electrode implanted in the inferior colliculus. In the first experiment, the experimental group was exposed to a low-level noise (an OBN centered at 0.5 kHz at 95 dB for 6 h/day for 10 days). After a 5-day recovery period when the thresholds returned to baseline, the animals were re-exposed to the same noise at 106 dB for 48 h. A control group was exposed only to the higher-level noise. The second experiment has an identical exposure schedule, except that the traumatic exposure involved a 4-kHz OBN presented at 100 dB. The 10-day "toughening" exposure reduced the amount of PTS in the low-frequency experimental group by 20–30 dB over the range of 0.5–4 kHz. Conversely, the amount of PTS in the high-frequency group was actually 15 to 25 higher than in the control group at and above the exposure frequency. The possible mechanisms involved will be discussed.

11:00

2PP8. A biophysical model of auditory-nerve responses to acoustic transients. David C. Mountain, Allyn E. Hubbard, and Mark Zagaeski (Dept. of Biomedical Eng., Boston Univ., Boston, MA 02215)

Acoustic transients are an important part of the natural acoustic environment as well as in acoustic communication signals such as human speech. The response to acoustic transients from a biophysically based simulation of the auditory periphery that includes models of cochlear mechanics, inner hair cell (IHC) receptor potentials, and neurotransmitter release at the auditory nerve (AN) afferent synapse were studied. The mechanical model is a 300-section, one-dimensional model of the basilar membrane (BM). Each section of the mechanical model drives an inner hair cell that includes a three-state transduction channel model and a two-state voltage-dependent channel model. The afferent

synapse models include an inactivating three-state calcium channel and a single compartment vesicle release model. The ensemble activity as a function of time for each level of the model (BM, IHC, AN) was visualized through the use of a color-raster display by representing the output amplitude of a given level with color. Transient stimuli were synthesized by adding together various simple signals: clicks, short-duration tone pips, exponentially damped sinusoids, and continuous noise. The transients produced distinctive spatial-temporal response patterns even in conditions of poor signal-to-noise ratios. This robustness of the spatial-temporal patterns appears to be due to a combination of the spectral representation created in the cochlear mechanics and the adaptation present in the auditory-nerve synapse. [Work supported by NIH.]

11:15

2PP9. Are high-frequency fibers necessary for speech perception in noise? E. A. Strickland, N. F. Viemeister (Dept. of Psychol., Univ. of Minnesota, Minneapolis, MN 55455), and D. J. van Tasell (Univ. of Minnesota, Minneapolis, MN 55455)

It has been proposed that auditory-nerve fibers whose CFs lie above the frequency region of speech may encode the periodicity of low-frequency signals, and may be used to perceive speech when the signal-to-noise ratio is low [S. Greenberg and W. Rhode, J. Acoust. Soc. Am. Suppl. 1 88, S23 (1990)]. If this is true, then it might be expected that identification of speech at low signal-to-noise ratios would worsen with the addition of a high-pass noise that is sufficiently intense to mask information in the high-frequency fibers. This hypothesis was tested for identification of vowels and spondees. Identification was measured as a function of signal-to-noise ratio in speech-shaped noise, with and without high-pass noise. The high-pass noise had a low-frequency cutoff of 3 kHz. The results indicate that the addition of high-pass noise does not degrade identification performance, regardless of the level of the speech-shaped noise, and also does not change the confusions made by listeners. These results suggest that the high-frequency channels are not necessary for speech perception under low signal-to-noise conditions or in quiet. [Supported by DC00110.]

TUESDAY MORNING, 30 APRIL 1991

INTERNATIONAL D, 9:00 A.M. TO 12:05 P.M.

Session 2SA

Structural Acoustics and Vibration: Structural Vibrations

Louis A. Herstein, III, Cochair

Tracor Applied Sciences, 2361 South Jefferson Davis Highway, Suite 1100, Heitman Center, Arlington, Virginia 22202-3863

Alan D. Stuart, Cochair

Applied Research Laboratory, Pennsylvania State University, University Park, Pennsylvania 16802

Chair's Introduction—9:00

Contributed Papers

9:05

2SA1. Comparison of continuous and discontinuous models of heterogeneity in the propagation of transient planar stress waves. Jerry H. Ginsberg and Xiang Xiao (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332)

In the characteristic differential equations governing propagation of linear one-dimensional waves through heterogeneous media, the only properties of significance are the sound speed c and the acoustic impedance ρc . The former occurs in the differential equations governing the (curved) characteristics, while the latter appears in the differential equations governing the evolution of particle velocity and strain along the characteristics. One might attempt to approximate the spatial variation of these material properties in a periodic array of homogeneous layers by smooth interpolating functions. An implicitly stable finite difference representation of the characteristic equations is used in the present work to consider propagation of a transient wave through a bi-material sequence of layers of thickness L . Waveforms are

compared to those obtained from a model in which the layer properties are taken to vary sinusoidally, such that $\rho = \rho_0[1 + \epsilon \sin(\pi x/L)]$ and $c = c_0[1 + \delta \sin(\pi x/L)]$. Of particular interest is the case where the wavelength is L , which, according to Floquet theory, represents a condition of uncertain stability for steady-state waves, but which leads to maximum interference for a harmonic transient wave in the sinusoidally varying medium.

9:20

2SA2. Finite element modeling of a radio ear bone vibrator. Emil R. Hayek, Martin J. Pechersky, and Alan D. Stuart (Graduate Prog. in Acoust., Penn. State Univ., P.O. Box 30, State College, PA 16804)

The Radio Ear B-71 bone vibrator is an electro-mechanical transducer used in audiologic tests to diagnose hearing disorders. Extensive studies have revealed that the bone vibrator exhibits a peaked low-frequency response that limits its clinical usefulness. The vibration of

this device has been previously characterized experimentally by optical holographic techniques. These results have indicated that the poor high-frequency response of the bone vibrator can be attributed to case resonances in the higher modes of vibration. To further investigate both the out-of-plane and in-plane vibration of the bone vibrator case, a finite element model was developed with the ANSYS system. The case mode shapes obtained previously by laser holography were used to verify the semiempirical FEM model. Once verified, this finite element model was used to analytically determine modifications to the bone vibrator that would improve its high-frequency response. This was accomplished by varying design specifications of the bone vibrator including its dimensions, stiffnesses, and masses and observing its effect on the case mode shapes.

9:35

2SA3. Soize's theory of structural fuzzy: An examination of fundamental assumptions. Victor W. Sparrow (Graduate Prog. in Acoust., 157 Hammond Bldg., Penn State Univ., University Park, PA 16802)

In 1986, C. Soize introduced a theory of scattering from structures that are known in gross aspects, but whose details are either unknown or only imprecisely known. Soize called his method a theory of structural fuzzy [Resh. Aérop. 1986-5, 23 and Resh. Aérop. 1986-5, 49 (1986)], which he applied to scattering in the midfrequency range. In the theory, Soize described complex mechanical subsystems by fuzzy finite elements, elements where the mass, rate of dissipation, and modal density are formulated using uniform probability density distributions. In this presentation, Soize's methods are outlined and the fundamental assumptions that underlay the theory are considered. Specifically, the implications of the uniform probability distributions and other assumptions on the fuzzy finite elements will be reported. Other probabilistic descriptions for these constituent elements may be more realistic. Theoretical limitations will also be summarized for the frequency ranges over which the fuzzy structure theory is predicted to be valid.

9:50

2SA4. Wave-number-frequency spectrum of turbulent wall pressure fluctuations. Carolyn Karangelen (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC 22064)

The study of the properties of turbulent wall flows is important to understanding the structural excitations that might induce flow noise. Models of the forcing functions of the structural excitations have been based on generalized properties of the spectral features of these flows. This paper will highlight the characteristics of turbulent spectral energy in wave number space, and present new experimental results. Discussion will address the wave number spectrum as a model for the forcing function of structural excitation. [Funded by ONR and IBM.]

10:05

2SA5. Optical measurements of shear waves on a point-driven cylindrical shell excited radially. Dowon Lee, Hyun-Gwon Kil, Christian Glandier, Jacek Jarzynski, and Yves Berthelot (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332)

A differential laser Doppler velocimeter (LDV) [Lee *et al.*, *Proceedings of the 3rd International Congress on Intensity Techniques*, Senlis, France, pp. 181-188 (1990)] has been used to measure in-plane motion associated with waves propagating on a thin cylindrical shell excited radially at a single frequency by a shaker. The endcaps of the shell approximate the case of a simply supported shell with axial constraints at both ends. The fiber optic technology used in the LDV system allows the optical probe head, located at a distance of about 30 cm from the shell, to be scanned over the entire surface of the shell. Data were recorded along an array of 32 points circumferentially and 16 points axially, enough to sample spatially both extensional (fast waves, low

wave numbers) and flexural (slow waves, high wave numbers) waves. The data were analyzed with both a two-dimensional spatial FFT algorithm and a Prony algorithm. The analysis clearly reveals both type of waves separately. Theoretical predictions are obtained from the analysis of a point-driven infinite shell and from Forsberg's original paper [AIAA J. 2 (12), 2150 (1964)]. Both theoretical and experimental results clearly show the presence of shear waves propagating on the structure. [Work supported by ONR.]

10:20

2SA6. Coupling of plate and shell modes at joints. I-Tai Lu, Henry L. Bertoni, and H. Y. Chen (Weber Res. Inst., Dept. of Elect. Eng., Polytechnic Univ., Route 110, Farmingdale, NY 11735)

For cylindrical and spherical shells without truncation or for flat plates with infinite lateral extent, fundamental shell or plate modes propagate independently without coupling. However, at a joint of two or more plate or shell elements, these modes couple together. If a mode is incident at an oblique angle from one element, it will excite other modes propagating away from the junction in each element. The propagation direction and the excitation strength of each mode can be determined by the following boundary conditions: continuity of displacement at the joint, continuity of rotation about the axis of the joint, and vanishing net force and torque on the joint. These boundary conditions are derived from the assumption that the joint is massless and has rigid cross section but offers no resistance to extension along and twisting about the axis of the joint, and to bending transverse to the axis. To be consistent with thin plate/shell theory, it is necessary to account for the first cutoff mode in addition to the three propagating modes in each plate or shell. Numerical examples will be discussed. [Work supported by ONR.]

10:35

2SA7. Alumer: A "rigid," acoustically transparent, composite material. Pieter S. Dubbelday and Forrest M. Eggleston (Naval Res. Lab., Underwater Sound Reference Detachment, P.O. Box 568337, Orlando, FL 32856-8337)

The elastic properties of foamed aluminum have been reported in an earlier presentation [J. Acoust. Soc. Am. Suppl. 1 88, S22 (1990)]. When the open-cell structure of this material is impregnated with a light elastomer, a composite material results that has a density and dilatational wave speed close to that of water. The Young's modulus and bulk modulus of this "Alumer" material have been measured, as a function of frequency, for various combinations of foamed aluminum, characterized by relative density and pore size, and polyurethane. The aluminum matrix imparts stiffness to the material, for use as an acoustically transparent backing plate or other structure. The elastomer may be chosen to give more or less absorptive properties to the material. [Work supported by the Office of Naval Research.]

10:50

2SA8. Modeling of impact response in moving chains. Sabih I. Hayek, Shyi-Ping Liu, K. W. Wang (Penn State Univ., University Park, PA 16802), and Francis H. K. Chen (General Motors Res. Labs, Warren, MI 48090-9055)

A significant contributor to noise from chain drives can be traced to the impact of a roller chain on a sprocket. A model is developed to estimate the impact impulsive load generated by an axially moving roller chain on a sprocket during the meshing process. Due to the dependence of the impulsive impact force on the velocity of the chain before impact, the impulsive load at each impact varies from impact to impact. The meshing process then produces nonperiodic and nonuniform impulsive impact forces. The model is used to analyze the response of the roller chain after each impact. The chain response model is being

developed to increase the understanding of the noise generated from engine drive chains. [Work supported by General Motors Corporation.]

11:05

2SA9. Behavior of multiple degree of freedom systems having neighboring eigenvalues: Veering of the eigenvalue loci and high mode sensitivity. Pei-Tai Chen and Jerry H. Ginsberg (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332)

Interesting phenomena occur when linear systems contain natural frequencies that become close as a system parameter is altered. The behavior of eigenfunctions and eigenvalues is studied by applying a perturbation analysis to a self-adjoint system. Veering of the eigenvalue loci versus the small perturbation parameter is observed and the criterion for the occurrence of veering is identified. It is proven that in the veering region, the eigenfunctions are highly sensitive to the value of the system parameter. A corollary is that in a system, whose individual subcomponents are highly coupled and nearly identical, eigenvalue veering and high mode sensitivity are associated with mode localization, in which some of the system modes are strongly enhanced in a small portion of the domain. An example of a previously studied two-span beam with irregular spacing of the supports displays the correlation and mode localization, the high mode sensitivity, and veering of the eigenvalue loci. Results of the perturbation analysis are shown to be virtually identical to those of the exact solution of the equations of motion. [Work supported by the Office of Naval Research, Code 1132-SM.]

11:20

2SA10. Natural frequencies and bending mode shapes of thin perforated plates using a variational method. Olivier Beslin, Pascal Millot (Laboratoire des Sciences de l'Habitat, Ecole Nationale des Travaux Publics de l'Etat, 69518 Vaulx-en-Velin, France), and Jean-Louis Guyader (Laboratoire Acoustique et Vibrations, Institut National des Sciences Appliquées, 69621 Villeurbanne, France)

A theoretical method to predict eigenfrequencies and mode shapes of a perforated plate was developed by building its Hamiltonian as a functional of the displacement field. The displacement functional base was chosen to be the natural base of a nonperforated plate having the same geometrical and mechanical characteristics as the perforated one with simply supported boundary conditions. Limiting the base vectors to a finite number, the extremization of the Hamiltonian was transformed into a Rayleigh-Ritz problem. Formulas of this analytical development were implemented on a computer to solve the eigenvalue problem for any kind of rectangular perforations. Numerical singularities were encountered, regarding the holes' shapes and sizes, and vector

sets. They were removed by using some simple physical considerations. The program developed for this study was tested by treating special cases, well known in the literature. For example, the case of a full plate with free boundary conditions was simulated by introducing a crown of holes around the edges of the plate, setting it free of its supports. Several cases of perforated plates were also computed. This work is the first part of a more general study on sound radiation from plates with holes and perforations.

11:35

2SA11. Matched asymptotic expansions solution for low-frequency sound radiation from fluid-loaded, baffled plates. Christiaan Kauffmann (Faculty of Technical Math. and Informatics, Delft Univ. of Technol., P.O. Box 356, 2600 AJ Delft, The Netherlands)

The method of matched asymptotic expansions is used to investigate sound radiation from fluid-loaded, baffled, thin elastic plates of dimensions small compared to the acoustic wavelength, i.e., for $k_0 a \ll 1$, where k_0 is the acoustic wave number and a is a typical size of the plate. In this low-frequency region both near-field quantities (plate response and surface pressures) as well as far-field radiation properties are obtained. At low frequencies, the plate radiates sound like a monopole, the strength of it being proportional to the net volume velocity of the plate. The consistency of the method is tested by power balance considerations that are shown to be valid up to leading order. [Work supported by the TNO Institute of Applied Physics, Delft, The Netherlands.]

11:50

2SA12. Input mobilities and power flows for edge excited, semi-infinite plates. Christiaan Kauffmann (Faculty of Technical Math. and Informatics, Delft Univ. of Technol., P.O. Box 356, 2600 AJ Delft, The Netherlands)

This paper relies highly on a paper by E. Eichler [J. Acoust. Soc. Am. 36, 344 (1964)], which contains an analysis of the response of a semi-infinite, thin plate excited by a time-harmonic load applied at a stud of width $2a$ attached to the edge, the remainder being free. Eichler used spatial Fourier transforms in the edge direction to obtain closed form integral expressions for the plate response. The present paper provides full details on the analysis (some of them not yet published) and presents new expressions for the input power fed into the plate. Numerical results covering a wide range of ka are presented, allowing for a simple physical interpretation in three cases: (a) $ka \ll 1$: point excitation, (b) $ka \rightarrow \infty$: line (or beam mode) excitation, and (c) $ka = O(1)$: interference effects at intermediate frequencies, which are also present in the simple acoustical analogy of a baffled piston. [Work supported by the TNO Institute of Applied Physics, Delft, The Netherlands.]

Session 2SP**Speech Communication: Issues in the Analysis of Articulatory Movement
(Lecture and Poster Session)**

Katherine S. Harris, Cochair

CUNY Graduate Center, 33 West 42nd Street, New York, New York 10036

Judith A. Cooper, Cochair

*National Institutes of Health, Federal Building 1C-06, 7550 Wisconsin Avenue, Bethesda, Maryland 20892***Chair's Introduction—8:00****Invited Papers****8:05****2SP1. Comparative investigations of speech and other motor behaviors.** Anne Smith (Dept. of Audiology and Speech Sci., Purdue Univ., West Lafayette, IN 47907)

This presentation focuses on the question of how articulator movements are controlled by the nervous system and on the question of whether neural control of speech movements is "special," that is, distinctive from the neural control processes underlying the production of other motor behaviors. To address these questions, muscle activity and movements have been recorded while subjects engaged in a variety of tasks, including speaking, finger tapping, mastication, and voluntarily controlled respiration. The results of these experiments suggest that there are common characteristics in the neural mechanisms underlying the control of speech and other motor behaviors. In addition, however, the experimental evidence suggests that sources of neural drive that appear to generate input to muscles for relatively automatic motor behaviors, such as mastication and respiration, are not equally dominant in controlling muscle activity for speech. Implications of these data for the concept of a "rhythm generator" for speech will be discussed. [This work has been supported by Grant DC00559 from NIH's NIDCD.]

8:30**2SP2. Prosodic categories and duration control.** Mary E. Beckman (Ohio State Univ., Linguistics, 204 Cunz Hall, Columbus, OH 43210-1229) and Jan Edwards (Hunter School of Health Sci., New York, NY 10010)

Two prosodic categories are particularly interesting for examining the relationship between phonetic units and articulatory timing because they provide a clear contrast in prosodic function: The durational correlates of "sentence stress" mark constituent peaks (nuclear-accented syllables), whereas phrase-final lengthening marks constituent edges. When kinematic patterns are contrasted in nuclear-accented versus unaccented and phrase-final versus nonfinal syllables, it seems that final lengthening is a local decrease in stiffness, whereas lengthening for accent is a change in the phasing of the final (consonant) gesture relative to the opening (vowel) gesture. When interaction with overall tempo is examined, however, it is clear that the prosodic function cannot be equated directly with the dynamic parameters, since at slow tempo, phrase-final lengthening affects phasing rather than stiffness. Thus prosodic structure is not the immediate goal of the dynamic representation. Rather, there must be intermediate representations of the phonetic tasks in terms of a local tempo change (for final lengthening) and an increase in the duration of the "sonority peak" of the syllable (for sentence stress). [Work supported by the NSF.]

8:55**2SP3. Experimental levels in the analysis of articulatory movements: Implications for speech motor control.** Vincent L. Gracco and Elliot L. Saltzman (Haskins Labs., 270 Crown St., New Haven, CT 06511)

Kinematic analysis of speech movements can provide important insights into the organization of underlying central processes. As such, much theoretical significance has been attached to movement variations hypothesized to reflect corresponding contrasts in motoric or linguistic goals. Too often, however, important properties of the physical plant and representation of the underlying control signals are ignored when interpreting such variation. Interpretation of movement without regard to the structure and biophysical properties of the controlled system and associated bioelectric events may result in overspecified models of

speech production. Further, organizational principles or neural control strategies for speech generated from kinematic observation without concomitant electromyographic observation may result in unrealistic theoretical abstractions. The utility of including, in a general sense, biomechanical and electromyographic levels of analysis in the development of detailed theories and models of speech production will be presented. Tentative speculations on characteristics of the speech motor control process will be discussed. [Work supported by NIH.]

9:20

2SP4. On the analysis of speech movements. John R. Westbury (Waisman Ctr., Univ. of Wisconsin, 1500 Highland Ave., Madison, WI 53705-2280)

New technological developments have made it possible to compile, in *one* afternoon, a speech kinematic data set that once required 3 months of tedious, exhausting effort. In a week's time, sets that once required a year, and in a month, sets that once required several years, can be compiled. To those old enough to have done things in the old ways, this is surely remarkable, but perhaps also unsettling because there are few sound traditions to guide the analysis of this rapidly expanding articulatory database. Heretofore, a central feature of many analysis schemes has been a heavy reliance on identifying "magic" moments in time and places in space—easily defined "events" in the kinematic signal streams. Articulator positions and velocities associated with these events, and time and distance intervals between them, then provided a parsimonious characterization of the movements. This analytic approach is certainly useful, but also troubling, first because the functional significance of the measurement events is unknown; second, because their locations are sensitive to the essentially arbitrary conventions that must first be established to represent the data; and finally, because the few discrete measures that are made may under-represent the phenomena of interest, and lead to flawed generalizations. Examples will be provided to illustrate these weaknesses of the classical approach, and partial remedies will be suggested. It is high time that we talk about methods.

9:45

2SP5. The degrees of freedom in controlling articulations. Keith Johnson, Peter Ladefoged, and Mona Lindau (Dept. of Linguistics, UCLA, Los Angeles, CA 90024-1543)

What do speakers control in the production of vowels? Data from two groups of five speakers saying words containing ten different American English vowels indicate that in saying the same vowel there is a great deal of variation between speakers in the articulatory gestures used, but considerable consistency within speakers in the articulations used. All ten speakers had constrictions in similar regions of the vocal tract for each vowel; but the relation between tongue movement and jaw movement was variable. Principal components analyses of the locations of six pellets showed that, for one group, two of the five subjects used the jaw and tongue in combination, and their vocal tract shapes could be described in terms of only two components. For the other three subjects, three components were needed to account for the same amount of variance, with the third component distinguishing tense and lax vowels. The other group of five subjects was analyzed separately. Two subjects showed good correlation between tongue height and jaw height; for the other three, tongue height was not well predicted from jaw height, except in the case of back vowels. These speakers used various combinations of articulatory mechanisms for separating tense and lax vowels. Lip positions were also examined. Even in the case of back vowels, only some of the speakers had good correlation between lip opening and protrusion. In conclusion, the evidence that vowel production is goal oriented is discussed using targets learned as auditory goals but produced as individually controlled articulatory gestures. [Work supported by NIH.]

10:10

2SP6. The control and kinematics of human jaw movement. David J. Ostry, J. Randall Flanagan, and Anatol G. Feldman (McGill Univ., Montreal H3A 1B1, Canada)

The kinematics of two-dimensional human jaw motion are presented based on x-ray microbeam recordings. The relationship between jaw translation and rotation is described and experimental records are compared with simulations based on the equilibrium point hypothesis (λ model). In general, jaw rotation and translation were found to start and end simultaneously and straight line paths were observed when rotation was plotted against translation. Several manipulations suggest that jaw rotation and translation are separately controlled. For example, when jaw movements in speech were examined, the slope of the relationship between rotation and translation varied with the consonant but did not depend on the vowel or speech rate. The kinematic patterns of jaw motion are well accounted for by the λ model. The model demonstrates that separate central commands can be defined associated with jaw translation, jaw rotation, and coactivation of muscles without motion. Central commands may be superimposed to produce combinations of rotation, translation, and coactivation. Empirical patterns can be captured by the model under the assumption of simple constant velocity shifts in equilibrium governed by central commands.

Poster Papers

All posters will be on display and all authors will be at their posters from 10:50 a.m. to 12:00 noon.

2SP7. Translating pellet positions into constriction features. Stanley Ahalt, Ashok Krishnamurthy, Tzzy-Ping Jung (Electrical Eng., Ohio State Univ., Columbus, OH 43210-1272), Mary E. Beckman, Kenneth De Jong, and Sook-hyang Lee (Ohio State Univ., Columbus, OH 43210-1229)

Tongue constriction features can be estimated from sagittal x-ray pictures of the tongue surface and vocal tract wall. However, such records cannot be obtained in quantity, making them unsuitable for testing models such as Stevens's quantal theory. The x-ray microbeam allows larger data sets, but records flesh points rather than surfaces. This paper presents an algorithm for relating the two representations. The vocal tract wall is estimated from whole-head scans and a palate trace. Pellet positions are then "warped" into a Cartesian space where location along the tract and distance from it are the x and y values. The algorithm has been applied in a replication of Perkell and Nelson's test of quantal theory using principal component analysis. Quantal theory predicts that the pellet closest to the constriction site will show least variability, and that the most precision will be in the dimension perpendicular to the vocal tract wall for "quantal" vowels such as /i/. In the warped space, then, the principal component of variation for the relevant pellet for these vowels should be parallel to the x axis. This prediction is borne out. [Work supported by the NSF.]

2SP8. An x-ray microbeam study of Hindi vowel articulations. R. Prakash Dixit and Raymond G. Daniloff (Dept. of Speech Commun. Disord., Louisiana State Univ., Baton Rouge, LA 70803-2606)

Using the x-ray microbeam system of the University of Wisconsin at Madison, articulatory movement data were recorded from pellets attached to the tongue, upper and lower lips, lower jaw, and velum while a native speaker of Hindi produced ten oral vowels contained in nonsense words of the form bVb. The data do not support the traditional descriptions of Hindi vowels given in terms of either tongue height or vocal tract openness. For example, overall position of the tongue dorsum was higher for /e/ than /i/, and similar for /u/ and /o/, and also for /a/ and /o/. Position of the jaw was lower for /o/ than for /a/. The lips were more open for /o/, /u/, and /u/ than for /o/, and more open for /i/ than /i/, /e/, and /e/. Position of the lips for /e/ and /e/ was identical. The velum showed somewhat greater elevation for /e/ than /i/, and identical elevation for /u/ and /o/. The remaining relations among various structures involved in vowel production were as expected. However, overall relationships among these structures were quite complex.

2SP9. A multisectional representation of the tongue surface based on ultrasound scans for time-varying vocalizations. Marc A. Cordaro (Dept. of Biomedical Eng., Johns Hopkins Univ., 144 New Engineering Bldg., Baltimore, MD 21218), Maureen Stone (National Inst. of Health, Bethesda, MD 20892), Moise H. Goldstein, Jr. (Johns Hopkins Univ., Baltimore, MD 21218), and Michael Unser (National Inst. of Health, Bethesda, MD 20892)

This paper describes a technique for producing a movie of multiple planes of the tongue surface based on midsagittal and coronal ultrasound scans of the tongue during time-varying vocalizations. For each scan view, the ultrasound images and acoustic signal of a given vocalization are recorded on videotape. The resulting sequence of video fields is digitized and processed to yield a sequence of two-dimensional coronal tongue profiles. Formant and fundamental frequencies are determined from the acoustic signal segment corresponding to each video field. Data for a number of coronal scan views are recorded with the vocalization repeated for each view. The resulting tongue profile sequences are time warped in a piecewise linear fashion (utilizing characteristics of both the ultrasound and acoustic data) to produce a single sequence of multiple time-aligned tongue surface profiles that are displayed in slow motion or freeze-frame modes.

2SP10. Objective evaluation of vowel pronunciation. Mark J. Bakkum, Reinier Plomp (Dept. of Oto-rhino-laryngology, Free Univ. Hospital, P.O. Box 7057, 1007 MB Amsterdam, The Netherlands), and Louis C. W. Pols (Univ. of Amsterdam, The Netherlands)

Usually pronunciation is evaluated subjectively by listening. The aim of this research project is to obtain an objective measure of the quality of pronunciation. All 15 Dutch monophthongs and diphthongs spoken by 24 males (deaf, foreign, and native Dutch norm speakers) were spectrally analyzed. The digital, real-time analysis system consisted of 16 bandfilters according to the critical-band model. Besides level and speaker normalization, time averaging was also applied so that all vowels were determined by a single point in a 16-dimensional spectral space. By principal components analysis, the number of dimensions was reduced, thus making a plain visualization possible. The feasibility of the objective evaluation has been investigated by considering to what extent the spectral information, as expressed in various distance measures, does give an adequate description of subjective judgments of the vowels, as obtained by experienced listeners. The overall scores (averaged over 15 vowels, after PCA) show a correlation coefficient of 0.97. Correlations per vowel are lower, but significant in nearly all cases. [Work supported by Netherlands Organization of Scientific Research (NWO) and Institute for the Deaf, Sint-Michielsgestel.]

2SP11. An x-ray microbeam study of the effect of lexical stress on the articulation of intervocalic velar stops. Alice Turk (DMLL, Morrill Hall, Cornell Univ., Ithaca, NY 14853-4701)

Previous research on two speakers showed that the closing and opening gestures of the upper lip during the production of intervocalic labial stops behave differently, depending on the stress of the following vowel [A. Turk, J. Acoust. Soc. Am. Suppl. 1 88, S56 (1990)]. When the consonant precedes a stressed vowel, its closing gesture (characterized by the measurement peak velocity/vertical displacement) is slower than its opening gesture. When the consonant precedes an unstressed vowel, its closing gesture is faster than its opening gesture. An analysis of velar stops will show whether the pattern observed for labials

can be generalized to another place of articulation. Results to be presented are of tongue movement during closing and opening gestures of the velar stops in the following words: /k/: *locale local fôllele*; /g/: *cigar vigor vinegar*. [Work supported by Sigma Xi.]

2SP12. Toward the development of articulatory signatures for intelligibility test words: Lingual kinematics. Gary Weismer, Ray D. Kent, and Greg Turner (Dept. of Commun. Disord. and Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705-2280)

The development of "normal" articulatory signatures for intelligibility test words should eventually permit highly specific designations of the articulatory basis of intelligibility deficits in persons with motor speech disorders. Our study of lingual kinematics for selected words, using x-ray microbeam data, suggests that a major obstacle to the construction of such signatures is the substantial interspeaker variability in pellet trajectories for a given word. This variability will be reported, ways in which it might be minimized will be discussed, and some analyses of relationships between lingual kinematics and vocalic formant trajectories will be described. One conclusion that has emerged from analyses to date is that typically used kinematic measurands such as **peak velocity, instantaneous acceleration**, and so forth, will not be very useful in the development of an articulatory signature. Rather, the unfolding of articulatory gestures over time must somehow be incorporated into the signature concept. [Work supported by NIH.]

2SP13. Linguo-mandibular coordination in consonant production. Alice Faber (Haskins Labs., 270 Crown St., New Haven, CT 06511) and Edda Farnetani (Centro di Fonetica del CNR, 35122 Padova, Italy)

The coordination between the jaw and tongue in regulating tongue position for consonant articulation was explored by simultaneous use of electropalatography and an alternating magnetic field device (Movetrack: Branderud, 1985). Linguo-palatal constriction was monitored via palatography, and tongue body and jaw height and frontness were monitored by small receiver coils adhering to the articulators. Speech materials were real and nonsense words of the form (C)V₁CV₂, where C ranged over {t d ʃ z l} and V₁ and V₂ over {a i}. The "real" words were produced in isolation and in sentences. Each token was repeated three times by one female native speaker of Italian. Regression analysis was used both to distinguish between rotational and translational movement of the jaw and to separate between the tongue's independent movement and positional effects caused by jaw movement. Examination of the inherent differences among consonants in tongue position reveals both synergistic and antagonistic patterns of linguo-mandibular coordination. The tongue body is lower for /z/ than for the other consonants, and fronted for (in order) /z ʃ/ > /t/ > /d l/. Comparison with the jaw positions and with the residuals from the regression analyses reveals that the tongue height differences result from a combination of jaw height differences (/z ʃ/ > /t d/ > /l/) and active lowering of the tongue body (relative to the jaw) for /z ʃ/. Similarly, the tongue frontness differences result from a combination of (translational) jaw fronting for /z ʃ/ and relative retraction of the tongue body for /l/. [Supported by ESPRIT ACCOR II BRA action 3279 and NICDC Grant No. DC-00016.]

2SP14. Effects of emphatic stress and speaking mode on articulatory organization. H. Timothy Bunnell and James Polikoff (Speech Processing Lab., A. I. duPont Inst., Wilmington, DE 19899)

An adult male talker recorded multiple repetitions of short nonsense sentences. Each sentence was of the form "A huC1 ate a C2uffle," where C1 and C2 were from the set {b/,d/,g/}. Sentences containing all nine combinations of C1-C2 pairings were recorded in several speaking modes: CLEAR, CONVERSATIONAL, STRESS1, and STRESS2.

The latter two conditions entailed production of the sentences in conversational mode, but with emphatic stress on the syllable containing either the first (STRESS1) or the second (STRESS2) variable consonant. These materials were recorded at the University of Wisconsin Microbeam facility and include both acoustic data and tracings of the trajectories of flesh points on the tongue, lips, and jaw. Analyses of the acoustic and articulatory data suggest that local effects of stress on consonant articulation are similar to those of clear speech. The largest acoustic and articulatory differences were observed for consonants in syllable final position. The various significant articulatory effects of stress and speaking mode will be compared to perceptual measures of consonant intelligibility.

2SP15. Mechanisms for tongue-tip trills. Richard S. McGowan (Haskins Labs., 270 Crown St., New Haven, CT 06511)

Possible mechanisms for the production of tongue-tip trills are discussed. It is necessary that energy be transferred from the airstream to the tongue tip to enable it to vibrate. A two-mass model, similar to that of the vocal folds, is one possible model for tongue trills, but there are others that allow the tongue one-degree-of-freedom motion. It is possible that the boundary-layer separation point is affected by the acceleration of air, which is not accounted for in the quasistatic approximation. An argument will be given that the acceleration term produces a favorable energy transfer during trills. Also, the capacitive loading of the vocal tract behind the tongue constriction can provide necessary phase shifts between constriction volume velocity and intraoral pressure for favorable energy transfer. A numerical simulation of the latter model is presented with parameters derived from measurements of oral-nasal volume velocity and intraoral pressure during sustained trills. Some of the features of the trills are reproduced with this simulation.

2SP16. Formant trajectories in the speech of normally articulating and misarticulating children. Karen Forrest, Ying-Chiao Tsao, Gary Weismer (Dept. of Commun. Disord. and Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705-2280), Mary Elbert, and Daniel A. Dinnsen (Indiana Univ., Bloomington, IN 47405)

In previous papers, it has been shown that analysis of obstruent spectra can be a powerful way to quantify and classify patterns of misarticulation in children. The focus on obstruent spectra fits in with the established idea that a problem with obstruent production is the primary reason why a child may be classified as having an articulation problem, but the lesson from adult articulatory disorders [such as apraxia of speech: see R. Kent and J. C. Rosenbek, *J. Speech Hear. Res.* 26, 231 (1983)] is that the prominence of obstruent articulatory problems should not obscure the fact that much of the nature of the articulatory deficit can be revealed by examination of vocalic formant trajectories. A formant trajectory analysis of five normal and five misarticulating children will be reported, and the way in which the analysis may bear on the characterization of childhood articulatory/phonological disorders will be described. [Work supported by NIH.]

2SP17. Production of final consonant deletion in black dialect. Gina Michou, Sandra Hamlet (Dept. of Otolaryngol. and Dept. of Commun. Disord. and Sci., Wayne State Univ., Detroit, MI), and Lewis Jones (Harper Hospital, Detroit, MI)

Deletion of certain final plosives is a phonological characteristic of American black dialect. This study reports physiological observations on the production mechanisms for such consonant deletion. Videofluoroscopic (x-ray) speech data were obtained for monodialectal speakers of black dialect, who were among subjects serving as normal controls in a study of head and neck cancer patients. Nonsense phrases of the form h ə CVC were spoken when cued by a precisely articulated live

voice verbal model, uttered by a speaker of General American dialect. Some speakers of black dialect "omitted" the final plosives in this situation. Insights on production mechanisms affecting vowel transitions and glottalization of final consonants are reported based on tongue and hyoid bone activity in the final VC transition. [Work supported by NIH, Grant No. CA-43838.]

2SP18. Timing and displacement in the articulation of prominence. Kenneth de Jong (Dept. of Linguistics, Ohio State Univ., Columbus, OH 43210-1229)

Macchi (1985) found that linguistic stress expands the degree of jaw opening in vowels and closing in obstruents. Edwards *et al.* (1991)

report that greater accentual prominence is achieved by initiating closing movements later relative to opening movements, thus allowing the jaw to reach a lower position. This paper relates the measures of prominence used in the two studies in a corpus of x-ray microbeam data. Syllables with labial and alveolar consonants and high and mid vowels varied in stress from full but unaccented to stressed (prenuclear accented) to most stressed (nuclear accented). Completed analyses of jaw movement show differences among all three stress levels in the amount of jaw lowering and raising. One subject, in addition, expands the duration of the more prominent items by changing the relative timing of opening and closing jaw gestures, suggesting that multiple strategies are used to increase prominence. The paper will also report on presently incomplete analyses of tongue-movement trajectories.

TUESDAY MORNING, 30 APRIL 1991

LIBERTY B, 8:25 A.M. TO 12:30 P.M.

Session 2UW

Underwater Acoustics and Acoustical Oceanography: Inverse Methods

James F. Lynch, Chair

Woods Hole Oceanographic Institution, Woods Hole, Massachusetts 02543

Chair's Introduction—8:25

Invited Papers

8:30

2UW1. Inverse methods and detection and estimation theory. Arthur B. Baggeroer (Depts. of Ocean and Electrical Eng., MIT, Cambridge, MA 02139)

Inverse methods have long been used to determine models for geophysical processes. These methods are closely related to the topics of parameter estimation and system identification in the detection and estimation theory literature, yet results from this area are seldom exploited in inversions. System concepts such as controllability and observability are important in specifying an inverse operator; threshold concepts are needed when parameters are nonlinearly related to observables and/or perturbations are employed; yet they are seldom used. More important, there are many bounds upon the performance of parameter estimators that can be applied to inverse problems in geophysics. This presentation will discuss how methods in system theory and detection and estimation can be applied to geophysical inverse problems. [Work supported by ONR.]

9:00

2UW2. Determination of geoacoustic parameters of the ocean bottom—Data requirements. Subramaniam D. Rajan (Dept. of Appl. Ocean Physics and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543)

An important problem in ocean acoustics is the determination of the acoustic parameters of the ocean bottom sediment layers. A variety of inverse methods has been proposed in the literature for obtaining these quantities from measurements of the acoustic field in the water column using either narrow-band sources. The ability of some of the perturbative methods to yield accurate estimates of the unknown parameters is investigated. For shallow water experiments, it is shown that a full wave method that uses the complex pressure field as data is nonlinear, the nonlinearity increasing with frequency and waveguide thickness. Methods that use modal eigenvalues as input data are only weakly nonlinear and can successfully yield estimates with acceptable resolution if the experiment is performed over a number of frequencies. In the case of deep water experiments, however, the experimental configuration can be so arranged as to make methods based on full wave inversion only weakly nonlinear.

2UW3. High-resolution matched-field inversion of ocean sediment parameters with simulated annealing. Michael D. Collins, W. A. Kuperman (Naval Res. Lab., Washington, DC 20375), and H. Schmidt (MIT, Cambridge, MA 02139)

High-resolution inversion of ocean sediment parameters is possible with matched-field processing. If the ocean bottom is complicated, matched-field inversion requires an efficient nonlinear optimization method such as simulated annealing [Kuperman *et al.*, J. Acoust. Soc. Am. 88, 1802 (1990)] to search the high-dimensional parameter landscape, which can have many local minima. An efficient propagation model is also essential because the wave equation must be solved many times. Single-frequency matched-field inversion simulations have been performed using various types of source and receiver arrays, including an array of sources beamed toward the ocean bottom, for problems involving fluid sediments, elastic sediments, and range dependence. The acoustic and elastic parabolic wave equations are used to construct replica fields for range-dependent inversion problems. For problems that can be regarded as range independent between the sources and receivers, large gains in efficiency can be achieved by working in wave number space using the synthetic aperture approach [G. V. Frisk and J. F. Lynch, J. Acoust. Soc. Am. 76, 205 (1984)] because the number of wave number samples required for inversion is much smaller than the number of wave number samples required to construct replica fields.

10:00

2UW4. An overview of the tomographic forward/inverse problem. Bruce D. Cornuelle (Scripps Inst. of Oceanography, Univ. of California at San Diego, La Jolla, CA 92093) and Bruce M. Howe (Univ. of Washington)

In ocean acoustic tomography, the travel time along a ray path is a weighted average of the propagation speed along the path. Because oceanographers generally have intuition about point measurements or simple averages, it has been necessary to transform travel time data into point values (maps) before communicating the results. The transformation can be done with a variety of methods, ranging from exhaustive Monte Carlo searches to Backus-Gilbert constrained estimation. The transformation converts travel time data with more or less independent errors to point value estimates with correlated errors that may have complicated, nonlocal structure. Since the error bars usually presented with an ocean map do not include the correlations, they do not accurately reflect the information content of a tomographic dataset. In addition, it is no longer possible to distinguish between the data errors and sampling blind spots by examining the error bars (or even the error covariances). Communicating tomographic results thus requires more effort, and more plots. Resolution (or averaging) kernels show how the estimate at a point is a weighted average of the entire field (with the average becoming more local as the number of rays increases). Null space vectors show fields that may be added to the estimated map without changing the data significantly. Given that the goal of ocean observations is to test dynamical hypotheses, it is also reasonable to consider transforming hypotheses into constraints on the travel times, rather than transforming the travel times into constraints on physical space hypotheses. [Work supported by ONR and ONT.]

10:30

2UW5. On the use of ocean dynamics to improve acoustic tomography estimates. Ching-Sang Chiu (Code OC/Ci, Dept. of Oceanography, Naval Postgraduate School, Monterey, CA 93943) and James H. Miller (Naval Postgraduate School, Monterey, CA 93943)

The resolution of maps obtained from ocean acoustic tomography is largely determined by the trajectories of the acoustic multipaths connecting sources and receivers. Since the distribution of crossings of the multipaths is nonuniform, tomographic resolution generally varies in space. Thus, depending on the characteristics of the sound channel and the scales of the ocean variability, "pure acoustic maps" can be inaccurate in those locations where spatial resolution is poor. In order to improve the resolution of the maps, it is necessary to add independent information to the inverse problem. In a computer experiment, the improvement in the tomography maps resulting from the incorporation of ocean dynamics is assessed. Here, the integration of dynamical information into tomography is accomplished using a Kalman filter. For the assessment, the maps obtained by assimilating synthetic tomography data into a nonlinear, quasigeostrophic ocean model are compared with the "pure acoustic maps." Moreover, the sensitivity of the Kalman filter output to the inexact specification of ocean dynamics is examined.

11:00-11:15

Break

11:15

2UW6. Converting bottom loss measured from a rough layered sediment to the equivalent "flat bottom" loss. Diana F. McCammon (Appl. Res. Lab., Penn State Univ., State College, PA 16804)

In the BLUG parameter-estimation technique, measured bottom loss is inverted to obtain a "best-fit" set of ten geoacoustic parameters that characterize the sediment. This inversion process has been improved and automated with Monte Carlo methods; however, in spite of these advances, the inversion process can still give poor results, notably in rough thin sediment regions, because the inversion model assumes flat interfaces between water, sediment, and basement. The purpose of this paper is to describe a correction factor that can be applied to the measured data to convert it from rough surface loss to the equivalent loss if the interfaces had been smooth. With this correction, the data can be made to conform to the assumptions of the model, which should lead the inversion process to a better fitting set of parameters. Four examples of this application are shown; in two thin sediment cases, the correction gave improved model/data correlations and lower squared errors; in the two smoother thicker sediment cases, the correction did not significantly affect the result. [Work supported by NAVOCEANO.]

11:30

2UW7. Time domain inversion of two-dimensional velocity fields using simulated annealing. P. Gerstoft, J. M. Pedersen, and P. D. Vestergaard^{a)} (Ødegaard & Danneskiold-Samsøe ApS, Kroghsgade 1, Copenhagen 2100, Denmark)

Inversion of scattered sound fields caused by horizontal and lateral velocity variations can be done using an efficient Monte Carlo method called simulated annealing. The subsurface contains large velocity variations in both depth and range and thus, for inversion of transient signals, at least a two-dimensional (2-D) representation of the velocity field is required. This 2-D description requires that the nonlinear inversion is carried out in a huge parameter space. Standard local optimization methods will be trapped in local minima and a search throughout is computationally prohibitive. Thus the simulated annealing method is used. The present implementation is here based on an ensemble approach whereby several copies are annealed simultaneously. By using several copies it is possible to obtain statistical information about the optimal cooling rate. In order to make the method converge in acceptable time, both optimization and the forward modeling method of the inversion have to be fast. Thus geometrically flexible but computationally exhaustive methods such as finite difference are not yet used. At present, the forward modeling is done by either the one-dimensional convolution model or ray tracing. The 2-D effect of the one-dimensional convolution is obtained by geometrically requiring the structure to be stratified with a weak variation in range. For the ray tracing, the substructure shall also be stratified but here the wave propagation is in a real 2-D environment. ^{a)}Presently at BIRPS, Cambridge, U.K.

11:45

2UW8. Linearization of the matched-field processing approach to acoustic tomography. A. Tolstoy (Acoust. Div., Naval Res. Lab., Washington, DC 20375)

This paper continues the approach presented in Tolstoy *et al.* [J. Acoust. Soc. Am. **89**, 1119-1127 (1991)] but offers a much improved

inversion technique, i.e., a linearization of the problem, which reformats the computations in terms of simple nonsquare matrix inversion for an overdetermined system. This linearization results in sound-speed accuracies that are an *order of magnitude* better than the earlier technique. In addition, calculations confirm that for simulations with white, Gaussian, uncorrelated noise, the linear/Bartlett processor results are identical to those of the minimum-variance/Capon processor. Finally, optimal source-receiver configurations have been determined by exhaustively computing the condition numbers for the associated matrices in the new linear formulation. Simulation results now show that *three* arrays located at optimal coordinates in a 250- by 250-km ocean region with shot sources distributed around the perimeter can result in 3-D sound-speed profiles determined to accuracies better than *0.07 m/s* and better than *0.03 m/s* for *four* arrays located at optimal coordinates. Such results presently assume perfect knowledge of sound-speed profiles at the arrays and around the region perimeter.

12:00

2UW9. Matched-field processing (MFP) tomography for inverting the El Niño profile. E. C. Shang and Y. Y. Wang (CIRES, Univ. of Colorado/NOAA/Wave Propagation Lab., Boulder, CO 80303)

The matched-field processing (MFP) method has been substantially applied for source localization studies in recent years. It has been demonstrated that the MFP is very sensitive to the mismatch of sound-speed profile (SSP) with a high-resolution MFP estimator. It turns out that the high-resolution MFP is also a potential powerful tool for SSP inverting with a known source-receiver system [A. Tolstoy and O. Diachok, J. Acoust. Soc. Am. Suppl. **1** **88**, S117 (1990)]. In this paper, the high-resolution mode matching (HRMM) estimator [E. C. Shang, J. Acoust. Soc. Am. **86**, 1960 (1989)] has been used for El Niño profile inverting. By matching a proper set of modal travel time perturbation, the El Niño profile can be efficiently inverted in a 2-D parameter space based on a simple acoustic model of the 1982-1983 El Niño event. [Work supported by ONR and NOAA.]

12:15

2UW10. Tomographic reconstruction of stratified fluid flow. Daniel Rouseff (The Johns Hopkins Univ. Appl. Physics Lab., Johns Hopkins Rd., Laurel, MD 20723), Kraig B. Winters, and Peter Kaczowski (Univ. of Washington, Seattle, WA 98105)

A new method for imaging a moving fluid using acoustic tomography is evaluated by numerical simulation. A cross section of the medium is probed by high-frequency acoustic waves from several different directions. It can be shown that the resulting measured travel time data contain sufficient information to reconstruct both the spatially varying sound speed and the transverse component of the fluid vorticity [K. B. Winters and D. Rouseff, *Inverse Problems* **6**, L33 (1990)]. The results are exact to within the validity of the straight-ray geometric acoustics approximation. To evaluate a discrete version of the reconstruction algorithm, a three-dimensional stably stratified mixing layer is simulated. The flow exhibits characteristic features in both density (sound speed) and vorticity. The dynamics of the fluid flow can be described as the instability of a vortex sheet. The acoustic travel time is calculated by integrating through the simulated flow fields. The synthetic data are then inverted to yield reconstructions of both the density and the vorticity of the evolving flow. [Work supported in part by the U.S. Navy under Contract No. N00039-89-C-0001.]

Session 3AA

Architectural Acoustics: Newer Measurement Procedures in Auditoria II

George E. Winzer, Chair

*Winzer Associates, 17721 Mill Creek Drive, Rockville, Maryland 20855**Invited Papers*

1:30

3AA1. Experience with early/reverberant sound energy ratios as a measure of auditorium acoustics. L. Gerald Marshall (KMK Associates, 7 Holland Ave., White Plains, NY 10603)

The advent of portable computerized instrumentation has made it practical for the acoustical consultant to collect extensive time/energy/frequency data during routine auditorium checkouts and to easily derive a number of potentially useful room acoustics measures from these data. The early/reverberant ratio is one measure that appears to correlate fairly dependably with the characteristics of clarity and reverberance for both speech and music. Experience with early/reverberant ratios will be discussed and measured data will be examined with respect to the variables of reverberation time, frequency, source and receiver locations, boundary reflectivity, and energy-ratio integration times.

2:00

3AA2. Room acoustic parameters with SLM and computer. Kjeld T. Hansen (Briel & Kjaer, 18 Naerum Hovedgade, DK-2850 Naerum, Denmark)

The Schroeder method, also called the integrated impulse-response method, is implemented using application module BE 7109 with an SLM (modular precision sound level meter B&K type 2231). A noise burst is generated by the SLM and fed into an amplifier and loudspeaker. The room response is then sampled and the reverberation times EDT, T(20), and T(30) are calculated automatically for 1/1-octave or 1/3-octave bands. The system gives very reproducible and accurate results which can be displayed. By connecting the SLM to a computer running the application program ROOMAC, which is part of BX 7109, the decay curves can be transferred to the computer. From these curves the more sophisticated parameters clarity, deutlichkeit, centre time, and total sound level can be calculated. These are displayed together with the sampled decay curves, the integrated curves, and the reverberation times after each frequency measurement thus giving the user immediate feedback. The results can be stored on disk. Furthermore, up to 12 positions can be spatially averaged. Altogether this makes the system especially valuable where a portable system and *in situ* calculation and diagnosis are needed. Measurement of the platform parameter SUPPORT will be discussed.

2:30

3AA3. Measurement of spatial information in auditorium by four closely located microphones. Yoshio Yamasaki (Sci. and Eng. Lab., Waseda Univ., 3-4-1, Okubo, Shinjuku-ku, Tokyo, 169 Japan)

Quite different impressions are felt in sound fields that have about the same reverberation time and sound pressure level. This kind of difference might come from the difference of spatial information, so it is important to grasp the spatial information in sound fields, especially of the early reflection periods. In this presentation, a way to grasp the spatial information of sound fields from impulse responses measured at four closely located points; the origin and three points of the same distance, 5 cm from the origin on the rectangular coordinate axes, is discussed. From these four impulse responses, the coordinates and powers of virtual image sources are calculated by correlation or intensity technique. Concert halls, opera theaters, and many other sound fields are measured by this technique. This measurement has been introduced not only to actual sound fields, but also to scale models. The distributions of virtual image sources and directivity patterns of auditoria and scale models are shown. These data bases are being used to estimate the sound fields and feed them back to acoustic design of auditoria.

3:00

3AA4. Room acoustics measurements using MLSSA. Douglas D. Rife (DRA Labs., 607 W. Nettletree Rd., Sterling, VA 22170)

The time-bandwidth product of a linear system is defined as the product of its bandwidth and its impulse-response duration. Due to its reverberation, the time-bandwidth product of a typical room is much larger than that of other common linear systems (e.g., acoustic transducers or electronic filters), thus posing special problems to the measurement system. It is shown that maximum length sequence (MLS) based measurements are inherently well suited to large time-bandwidth product systems, including auditoria. The MLSSA measurement system represents the first commercial implementation of MLS measurement methods and provides a variety of post-measurement processing functions for room acoustics applications.

3:30

3AA5. Room reverberation measurement by conventional methods and by the MLSSA measurement system. Mendel Kleiner and Peter Svensson (Dept. of Appl. Acoust., Chalmers Univ. of Technol., S-41296 Gothenburg, Sweden)

Experiments have been made to compare the measured T60 values measured by conventional analysis equipment to those measured by equipment using the maximum length sequence (MLS) correlation technique such as the commercial MLSSA system. Experiments have also been done using MLSSA impulse response results in conjunction with signal postprocessing in MatLab. Results show that the MLSSA system yielded values which differed by up to 10% from those obtained by conventional techniques. The reasons for this are not entirely clear. MatLab postprocessing also allows signal postprocessing to improve the dynamic range of measurement and the implementation of routines for more specialized room acoustic measures. [Work supported by the Swedish National Council for Building Research.]

TUESDAY AFTERNOON, 30 APRIL 1991

PRATT A, 3:00 TO 5:15 P.M.

Session 3MU

Musical Acoustics: Physics of Music: Measurements and Models

William J. Strong, Chair

Physics Department, Brigham Young University, Provo, Utah 84602

Contributed Papers

3:00

3MU1. Study of bowed string motions with optical detectors. Roger J. Hanson, Lawrence L. Dirkes, and Andrew J. Schneider (Physics Dept., Univ. of Northern Iowa, Cedar Falls, IA 50614-0150)

Bowed string phenomena have been studied on a system in which both ends of a string are mounted rigidly to a steel beam. Waveforms are studied at the "nut" and the "bridge" ends with photon-coupled interruptor modules (PCIM) [R. J. Hanson, *Phys. Teacher* **25**, 165-166 (1987)]. At positions farther from the ends, where the amplitude of string motion is greater than the linear region of the PCIM, one-dimensional photodiode arrays are used in an arrangement modified from that reported earlier [T. Nakamura, L. L. Dirkes, and R. J. Hanson, *J. Acoust. Soc. Am. Suppl. 1* **86**, S16 (1989)]. Emphasis is on the study of string motions with periods much greater than any natural period of the string which can be obtained with bowing forces greater than the Schelling maximum [R. J. Hanson, A. J. Schneider, and F. W. Halgedahl, *J. Acoust. Soc. Am. Suppl. 1* **86**, S15-S16 (1989)]. The

waveforms observed at different positions have been used to investigate the triggering of this motion with a period longer than that of the normal slip-stick bow and string interaction triggered by the Helmholtz "kink."

3:15

3MU2. Measurements of the impedance presented to the string by a bow. Bernard Ricca and Gabriel Weinreich (Randall Lab. of Physics, Univ. of Michigan, Ann Arbor, MI 48109)

It is well known that the various parameters involved in bowing a stringed instrument (e.g., bow "pressure") are important in determining the wave shape and stability of the vibrating string (and hence the sound produced by the instrument). A fuller understanding of this process, however, requires knowledge of the force characteristic of the bow on the string. The motion of a string vibrating both freely and while interacting with a bow have been measured and from these measure-

ments the impedance presented by the bow to the string was determined. The method for measuring string motion is based on earlier work [B. Ricca, G. Weinreich, and N. R. Michael, *J. Acoust. Soc. Am. Suppl.* 1 86, S15 (1988)]. By it, the impedance presented by the bow to the string both parallel and perpendicular to the bow hair is determined. Some results will be presented. [Work supported by NSF.]

3:30

3MU3. Violin admittance measurements using a one-dimensional mass-loading technique. Levon L. Yoder (Dept. of Physics, Adrian College, Adrian, MI 49221)

The input admittance of a violin can be determined from load/no-load measurements of the vibration velocity of the bridge [Weinreich and Yoder, *J. Acoust. Soc. Am. Suppl.* 1 81, S83 (1987)]. In the most successful of these experiments [Boutillon, Weinreich, and Michael, *J. Acoust. Soc. Am. Suppl.* 1 84, S179 (1988)], the load was a mass attached directly to the bridge. Since the bridge motion is three dimensional, the load impedance is also. The problem involved a 3×3 admittance matrix and required nine load/no-load measurements. This was accomplished by using three different acoustic irradiating fields and measuring the vibration velocities in each of the three directions. This paper presents a method of applying a one-dimensional load to the bridge by attaching a cylindrical mass to one end of a long sewing needle and immersing it in an air column to form a surrounding air cushion and to hold it upright. The data acquisition and analysis are reduced to three independent problems requiring only one acoustic field and separate mass loading for each direction. The admittance can be studied as a problem in one, two, or three dimensions. Experimental results in two dimensions will be presented.

3:45

3MU4. Vibrational modes of piano soundboards. Hervé Brelay^{a)} and Thomas D. Rossing (Dept. of Physics, Northern Illinois Univ., DeKalb, IL 60115)

The vibrational modes of the soundboards have been compared in two small upright pianos: one of solid spruce and one of three layers (two thin layers of spruce with a core of poplar). The model frequencies are quite similar, although the mode shapes are somewhat different. In the spruce soundboard, the bending wave velocities for the low-frequency modes appear to increase with the square root of frequency, which suggests that the stiffness of the soundboard plus the ribs is approximately the same along and across the grain (i.e., the soundboard is fully compensated). In the layered soundboard, on the other hand, the nodal lines tend to follow the ribs, indicating that the soundboard is overcompensated. Mode damping does not appear to be substantially different in the two soundboards.^{a)}Exchange student from Ecole Nationale Supérieure des Télécommunications, Paris.

4:00

3MU5. Acoustics of a yangqin. Jianming Tsai and Thomas D. Rossing (Dept. of Physics, Northern Illinois Univ., DeKalb, IL 60115)

The *yangqin* is a Chinese hammered dulcimer with a trapezoidal soundboard. It is played with two bamboo hammers and is used as a solo instrument as well as in Chinese traditional music ensembles. The instrument we studied, approximately 1 m by 0.5 m, has 54 notes covering a 4-octave range G_2 to G_6 (98–1568 Hz). It has five bridges and a total of 125 strings with sliders and rollers for fine tuning and for rapid modulation. The soundboard, which is crowned to a height of 4 cm in the center, is supported by seven unequally spaced transverse ribs. The ribs also divide the body into eight air chambers, which are connected by four or five holes (2.8 cm in diameter) through each rib; it is not clear whether these air chambers play an important role in the acoustics of the *yangqin*, however. Vibrational modes of the soundboard in the frequency range 100–700 Hz have been studied by impact modal anal-

ysis as well as by scanning with an accelerometer as the soundboard is driven by a small shaker. Nodal lines tend to follow the stiff transverse ribs. Model shapes indicate that the transverse stiffness is substantially greater than the longitudinal stiffness in the braced soundboard. The impedance at most points on the bass bridges shows a maximum around 100 Hz and then falls off at roughly 6 dB/octave. The impedance on the treble bridges, on the other hand, reaches a broad maximum around 2 kHz and falls off quite slowly with frequency, at least up to 5 kHz.

4:15

3MU6. Numerical and physical experiments on hammer and piano strings. Antoine Chaigne (TELECOM Paris, 46 Rue Barrault, 75634 Paris Cedex 13, France), Anders Askenfelt, and Erik V. Jansson (Dept. of Speech Commun. and Music Acoust., Royal Inst. of Technol. (KTH), P.O. Box 700 14, S-100 44 Stockholm, Sweden)

Numerical experiments have been made on the piano string with a discrete model using a finite difference calculation procedure. The string model was previously applied to the guitar [A. Chaigne, *J. Acoust. Soc. Am. Suppl.* 1 88, S188 (1990)] and is now tested with experimentally obtained data for the piano. The numerical experiments include systematic variations of parameters such as hammer velocity and nonlinearity of the felt hammer and their influence on generated waveforms and spectra. Earlier physical experiments on pianos have provided data on the velocity of the hammer and the string. In addition, the hammer-string interaction force has been estimated by measuring the compression of the hammer felt during string contact. Recorded string tones and measured waveforms will be presented and compared with typical examples of those calculated.

4:30

3MU7. Numerical modeling of guitar radiation fields using boundary elements. Matthew Brooke and Bernard E. Richardson (Dept. of Physics, UWCC, P.O. Box 913, Cardiff CF1 3TH, Wales, U.K.)

Boundary element (BE) methods have been used to supplement an existing numerical model of the classical guitar. This model computes, from fundamental parameters, the transfer between the plucking force at a point on the string and the acoustic pressure at the listening point. Structural mode shapes are determined from finite element (FE) analysis. The computed surface velocities are then used to determine the coupling between the top plate and strings and also the coupling between the top plate and Helmholtz air resonance of the body cavity. The resultant acoustic radiation is calculated using the BE formulation, in which the entire surface of the guitar body is divided into boundary elements with surface velocities being specified over the top plate and zero elsewhere (to simulate "rigid" back and sides). The computed radiation fields will be compared with measurements on real systems. Future work will involve coupling of the FE and BE methods in order to account for fluid loading.

4:45

3MU8. Time domain simulation of flute-like musical instruments. B. Fabre (Laboratoire d'Acoustique Univ. Paris VI, Tour 66, 4 Pl. Jussieu 75005 Paris, France and Univ. du Maine, Le Mans, France) and A. van Steenberghe (Eindhoven Univ. of Technol., 5600 MB Eindhoven, The Netherlands)

The time-domain description of musical flute-like instruments as proposed by McIntyre, Schumacher, and Woodhouse [*J. Acoust. Soc. Am.* 74, 1325–1345 (1983)] has not yet been fully exploited. A major problem is the very slow transient behavior of the model. For such models, based on the analysis of Fletcher [*J. Acoust. Soc. Am.* 60, 926–936 (1976)], the nonlinearity is only due to the jet drive saturation. The transient behavior is linear because this nonlinearity is "smooth" and does not affect low amplitude oscillations. Linear theory is most

useful as it allows an analytical solution of the problem. Comparison between the analytical solution and the numerical results indicates that the discretization can significantly affect the oscillation behavior. This effect is demonstrated for a simple flute model.

5:00

3MU9. Tuning of the Tongan nose flute *fangufangu*. Peter L. Hoekje (Dept. of Physics, Northern Illinois Univ., DeKalb, IL 60115)

The *fangufangu* is made from a cylindrical section of bamboo closed by two adjacent nodes. Bore radii (a) range from 1.5 to 5 cm and lengths (L) from 20 to 60 cm. Typically, six identical holes are burned

through the side walls: one at either end, two in the center, and two symmetrically placed near ($L/4$) from the ends. Hole radii (b) range from 0.5 to 0.9 cm. One end hole is blown through a slit between the bamboo wall and the player's nostril. The four traditional fingerings use the next hole and the farthest hole. All have two resonances within the lowest octave, but one mode does not sound and two are only about 50 cents apart. There are thus six playable notes in this range, but scale tunings vary widely among existing instruments. Multiple mode cooperations are rare. For a given value of (a), increasing (b/a) overall raises frequencies of most resonances in parallel, but one frequency rises faster than the trend and another falls. They cross when (b/a) = 0.2 (a = 1.6 cm, L = 48 cm). Individual hole sizes and positions can be perturbed for specific tuning effects, as shown by calculation and experiment.

TUESDAY AFTERNOON, 30 APRIL 1991

INTERNATIONAL E, 1:30 TO 3:55 P.M.

Session 3NS

Noise: Value of Industrial Audiometry

Julia D. Royster, Cochair

Environmental Noise Consultants, P.O. Box 30698, Raleigh, North Carolina 27622-0698

Alice H. Suter, Cochair

Alice Suter and Associates, 1657 River Dee Court, Cincinnati, Ohio 45230

Chair's Introduction—1:30

Invited Papers

1:35

3NS1. Industrial audiometry: Benefit or liability? Alice H. Suter (Alice Suter and Assoc., 1657 River Dee Ct., Cincinnati, OH 45230)

In the U.S., audiometry is usually considered an essential component of a hearing conservation program and it is widely used to identify workers whose hearing threshold levels have begun to deteriorate so that interventions can take place. Reliance upon audiometry may not always be warranted, however, especially when employers are tempted to skimp on their engineering noise control programs. There is a strong voice against requiring industrial audiometry among hearing professionals in Canada and Australia. These experts claim that the reliability and validity of industrial audiometry tends to be poor, that workers are often falsely reassured that management is taking proper care of their hearing, and that workers rights are being abridged by mandatory audiometric testing. Although these arguments are not always supported by this author, they are potentially important and worthy of exploration.

2:00

3NS2. Audiometric variability. Lynne Marshall (Naval Submarine Med. Res. Lab., Box 900, Groton, CT 06349)

The size of detectable hearing shifts for individuals in hearing conservation programs is limited by audiometric variability. This talk describes how different audiometric test methods affect threshold, both in terms of accuracy (i.e., the absolute value of the threshold) and precision (i.e., the variability of the threshold). Factors to be discussed include the general test procedure, the number of responses required to estimate threshold in clinical test procedures, the step size used to adjust levels for threshold measurement, and earphone replacement. Data from computer simulations, laboratory tests, and industrial hearing conservation programs will be discussed. [Work supported by VA-DOD.]

3NS3. Motivational benefits of audiometric monitoring. Rena H. Glaser (Med. Dept./3M, 3M Ctr., Bldg. 220-2E-02, St. Paul, MN 55144-1000)

Audiologists and other professionals who administer occupational hearing conservation programs have for years used audiometric test results to determine the effectiveness of their programs and to show management any departments or areas that require attention. One aspect of the value of audiometry in industry that has received minimal attention is the educational and motivational impact that audiometric test results may have on the employee. Ideally, the test results should be given to the employee immediately following the test. This gives the employee the opportunity to ask questions and allows the examiner to praise the employee's good hearing conservation practices if no significant changes are noted in the employee's audiogram. If hearing thresholds have decreased, the tester is given the opportunity to reinforce the need for hearing protection use both on and off the job. All employees who receive audiograms should receive a meaningful summary of their test results. If this summary cannot be done immediately following the audiogram, written feedback is imperative. Employees are more willing to cooperate with management in these programs when employees can ascertain a benefit to themselves.

2:50

3NS4. Industrial audiometry: Do we ask too much of it? M. E. Roberts (Hearing Conserv. Sec., Workers' Comp. Board, Box 5350, Vancouver, B.C. V6B 5L5, Canada)

As of 1978, industrial health and safety regulations in British Columbia have required annual hearing tests for workers exposed to eight hour-equivalent noise exposures of 85 dBA or greater. Industrial audiometric technicians are trained and certified by the WCB. In addition, the regulations require that facilities be inspected annually to ensure calibration procedures are used and background noise levels are monitored. All results are submitted to a central data bank. Audiometry and the extensive counseling that takes place at the time of the test have been useful in motivating workers to consider noise as a personal health hazard, particularly for those working in the lower range of hazardous levels. In addition, averaged audiograms for workers in specific occupations have been useful in predicting future claims costs and thus have also been an economic motivator for establishing and maintaining hearing conservation programs.

3:15

3NS5. Using audiometric database analysis results to prevent occupational hearing loss. Julia Doswell Royster (Environmental Noise Consultants, Inc., P.O. Box 30698, Raleigh, NC 27622-0698) and Larry H. Royster (NC State Univ., Raleigh, NC 27695-7910)

When audiometric monitoring results for individuals are used to identify employees whose hearing changes indicate inadequate protection from noise exposure, threshold shifts must occur before actions are triggered to increase protection for the affected person. In contrast, when audiometric results for groups of noise-exposed employees are evaluated using audiometric data base analysis (ADBA), the presence of high variability in threshold measurements across the population alerts hearing conservation program (HCP) personnel to program deficiencies before many individuals develop significant threshold shifts. This warning allows HCP personnel to improve the program, thereby preventing hearing loss. Several case histories of successful ADBA applications will be detailed to illustrate how ADBA results can be used to assess program effectiveness for different noise exposure groups, to answer specific questions about hearing protector adequacy, and to set priorities on noise control projects. The preventive benefits from applying the ADBA procedures recommended by ANSI S12 Working Group 12 provide a strong justification for the audiometric phase of the HCP. With more effective programs, reduced employer liability for occupational hearing loss is expected.

Contributed Paper

3:40

3NS6. Single and combined effects of noise, whole body vibration, ambient temperature, illumination, and different work tasks on changes in experienced subjective stressfulness. Olavi J. Manninen (Dept. of Environmental Health Sci., School of Hygiene and Public Health, Div. of Toxicological Sci., Johns Hopkins Univ., 615 N. Wolfe St., Baltimore, MD 21205)

Presently, there is a lively discussion about how to develop and design new exposure standards for the combined effects of noise and vibration. In order to provide a sound basis for this kind of standardization work and to reach a better understanding of the interactions

between the combined effects of the environmental factors involved, seven different experiments were carried out in a special exposure chamber. The studies focused on characterizing the changes in experienced subjective stressfulness. The experiments were either based on block design or factorial experimental design and 192, 80, 10, 15, 90, 108, or 60 healthy male volunteers participated. Results showed that the scores of subjective stressfulness ratings varied greatly depending on the type of the exposure combination and the stressfulness increased with the duration of the exposure. It appeared that exposure situations involving simultaneous noise (85, 90, 95 dBA) and sinusoidal 5-Hz whole body vibration (Z axis; 2.12 m/s^2 , 2.44 m/s^2) were rated more stressful than

noise or vibration alone. An elevated environmental temperature (30, 35 °C) increased stressfulness in situations where the subjects were exposed to stochastic vibration (2.8–11.2 Hz) and noise. In particular, young persons (19–25 yr)—in comparison to middle-aged (26–39 yr) or older (40–54 yr)—rated the additional stress caused by noise higher

when they were in a competitive situation or doing physically strenuous work and were simultaneously exposed to a whole body vibration at 35 °C. The illumination level of lighting together with smoking seemed to affect the experienced stressfulness, too. [Work supported by the Acad. of Finland.]

TUESDAY AFTERNOON, 30 APRIL 1991

LIBERTY A, 12:55 TO 5:00 P.M.

Session 3OC

Acoustical Oceanography: Open Workshop on Remote Sensing of Sediment Properties by Measurements in the Water Column

Robert D. Stoll, Chair

109 Oceanography Building, Lamont–Doherty Geological Observatory of Columbia University, Palisades, New York 10964

Chair's Introduction—12:55

Invited Papers

1:00

3OC1. Chirp sonar for classification of marine sediments. Lester R. LeBlanc (Florida Atlantic Univ., Boca Raton, FL 33431), Larry A. Mayer (Dalhousie Univ., Halifax, Nova Scotia B3H 4J1, Canada), and Steven G. Schock (Florida Atlantic Univ., Boca Raton, FL 33431)

The chirp sonar is a calibrated wide-band digital FM sonar that provides quantitative, high-resolution, low-noise subbottom data. In addition, it generates an acoustic pulse with special frequency domain weighting that provides nearly constant resolution with depth. The chirp sonar was developed with ONR funding to support the objective of remote acoustic classification of seafloor sediments. In addition to producing high-resolution images, the calibrated digitally recorded data are processed to estimate surficial reflection coefficients as well as a complete sediment acoustic impulse profile. In this paper, the focus is on the analysis of surficial sediments in Narragansett Bay, RI. Quantitative acoustic returns from the chirp sonar are used to estimate surficial acoustic impedance and to predict the sediment type. The estimates are compared to “ground truth” values determined from core samples. The comparisons show a high correlation between the core-determined sediment type and the estimates from the acoustic measurements. In concert with the field work, sediment classification models for various depositional environments are being developed. A robust model for deep-sea carbonates has been completed and more generalized models are now under development that predict impedance, density, porosity, compressibility, and rigidity. [Work supported by ONR.]

1:20

3OC2. Determination of compressional wave speed profiles using modal inverse techniques in a range-dependent environment in Nantucket Sound. George V. Frisk, James F. Lynch, and Subramaniam D. Rajan (Dept. of Appl. Ocean Physics and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543)

An inverse method for determining geoacoustic properties in a horizontally stratified, shallow-water waveguide is extended to the case of a weakly range-dependent environment [Frisk *et al.*, J. Acoust. Soc. Am. 86, 1928–1939 (1989)]. The technique consists of estimating the local modal eigenvalues from the beam-formed output of a horizontal array and using these data as input to modal inverse methods for obtaining the local bottom parameters. Specifically, the approach is applied to data at 140 and 220 Hz obtained in a shallow-water environment with a known abrupt change in bathymetry. First, a range-independent medium is assumed and both iteration of forward models and perturbative inversion methods are applied to the modal data to obtain estimates of the bottom sound velocity profile. Although the perturbative inversion results are clearly superior, neither approach reproduces the full dependence with range of the observed pressure fields or the complete modal peak structure. In particular, the data exhibit an apparent splitting of the modal peaks that is interpreted, within the context of adiabatic mode theory, as the superposition of the modal contributions from the two segments of the waveguide with differing depths. When the assumption of a range-independent medium is relaxed and the perturbative inverse technique is applied to the eigenvalue data from each portion of the waveguide, distinctly different bottom profiles are obtained for each part. These result in significantly improved agreement between theory and experiment, thereby demonstrating the effectiveness of the range-dependent inversion procedure.

3OC3. Estimates of physical properties of deep ocean sediments using deep-towed multichannel seismoacoustic data. J. F. Gettrust and M. M. Rowe (Naval Oceanographic and Atmospheric Res. Lab., Code 362, Stennis Space Center, MS 39529)

Data taken with a deep-towed acoustics/geophysics system (DTAGS) are used to study deep ocean sediments. By towing the 250- to 650-Hz bandwidth source and 48-channel hydrophone streamer within 400 m of the seafloor, spatial resolutions on the order of ~ 10 m with signal penetration to ~ 600 -m depth was achieved. Estimates of the interval compressional velocity (V_p) within the sediments generally are within several percent of "true" interval velocities. The V_p vs depth functions for both Atlantic and Pacific sites suggest that the V_p gradient is generally extremely low within the upper ~ 50 – ~ 125 m of sediments, then increases with depth; this result is not consistent with simple sediment compaction models used to develop previous geoacoustic models. Further, the spatial variability of V_p within the sediments ($> 5\%$ over distance scales of 10 to 100 s of m) resolved with these data indicates that range independent geoacoustic models are not satisfactory. Shear velocity estimates are also presented and used to constrain geoacoustic predictions. [Work supported by ONR and ONT.]

Contributed Paper

2:00

3OC4. Remote sensing of seabed compressional wave attenuation from acoustic cw propagation experiments combined with the bottom shear modulus profiler (BSMP) database. Andrew Rogers (Geoacoustics Lab., Univ. of Miami RSMAS (UM/GAL), Miami, FL 33149), Tokuo Yamamoto (UM/GAL), Fred Tappert (UM RSMAS), and William Carey (NUSC)

As a result of recent ONR shallow water ARI, an accurate database of the compressional wave velocity, the shear wave velocity, and the density of the sediment for the continental shelves of Northeastern U.S. is available (Yamamoto and Trevorrow, in this conference). It has been shown that the model structures of acoustic waves propagating in these

shallow waters are very accurately modeled using the BSMP database as compared with the cw propagation experiments [J. Acoust. Soc. Am. Suppl. 1 88, S106 (1990)]. In this paper, it is shown that the compressional wave attenuation within the seabed can be accurately determined from the wave-number energy spectra of cw propagation data from range-independent environments. The seabed geoacoustic data so obtained are then used in the modeling of acoustic cw propagation in range-dependent cases using Fred Tappert's PE code and compared with Bill Carey's experiments. Excellent agreements between model calculations and experiments are obtained, indicating the accuracy and stability of the combined BSMP-cw experiments for determination of seabed compressional wave attenuation in shallow water. [Work sponsored by ONR.]

2:15–2:30

Break

Invited Papers

2:30

3OC5. Upper crustal properties determined from wide aperture towed array data. John Diebold (Lamont-Doherty Geological Observatory, Palisades, NY 10964)

Marine seismic reflection/refraction data with unusually large apertures can be collected using two or more suitably equipped ships. Although multiship experiments have classically been designed to delineate deep structure and seismic velocity, the data are, under certain conditions of geometry and sediment/crustal character, useful for the investigation of shallow properties. Three examples broadly define the range of application. Large aperture (0–10 km) CDP measurements in the shallow waters over the thickly sedimented continental margin of the northeast U.S. included first arrivals corresponding to diving rays refracting in the sedimentary column. Straightforward traveltime analysis of these arrivals yields detailed 2-D P -wave velocities from the seafloor down to 3 km, except where velocity reversals are encountered. Data from a similar experiment over the East Pacific Rise, with 0- to 6-km offsets, can be analyzed for crustal velocities as shallow as 100 m below the thinly sedimented seafloor. In the Hudson River, sediments are sufficiently thin and the water shallow enough that upper crustal refractions are seen as first arrivals in single-ship CDP data. In this case, however, sedimentary properties can only be estimated by indirect methods. In each of these three data sets, some combination of velocity structure and experimental geometry allows velocity determination for upper sediments and crust.

3OC6. Acoustic sensing of the upper crust. Orest I. Diachok (Naval Res. Lab., Washington, DC 20375)

Acoustic interaction with the ocean bottom is dominated by the volcanic upper crust in virtually the entire Pacific and large portions of the Atlantic Ocean. Previously reported measurements by numerous investigators revealed very low sound speeds (~ 2500 m/s) in recently formed crust, compared to ultrasonic measurements in dredged hand held samples (~ 6000 m/s between 0–60 m.y.), hypothetically due to high large scale crack density. Recent measurements of the coherent component of the reflection coefficient in 40–60 m.y. old, nearly sediment-free (< 40 m) crust, provide accurate measures of critical angles, and by inference, compressional and shear speeds. In the absence of fracture zones, the estimated compressional sound speeds in the older crust are generally only slightly greater than in young crust. Shear speeds at some sites in this age range exceed V_w , the speed of sound in water, whereas in young crust $V_s < V_w$. As a result, at sufficiently low frequencies and small grazing angles, where roughness effects are minimal, reflectivity is low (near zero when $V_s \lesssim V_w$) in young crust and may be near unity in 40–60 m.y. old crust. Implications of these results for acoustic field prediction and future research in acoustic sensing of the crust will be discussed. [Work supported by ONR.]

3:10

3OC7. Seismo-acoustic waves for remote sensing in shallow-water sediments. Tuncay Akal (SACLANTCEN, Viale S. Bartolomeo 400, 19026, La Spezia, Italy) and Robert D. Stoll (Lamont-Doherty Geological Observatory of Columbia Univ., Palisades, NY 10964)

The propagation and attenuation of seismic and acoustic waves observed in shallow water are strongly dependent on the physical characteristics of the seafloor. Because the acoustic field in the water column and seismic propagation in the seafloor are directly related, different techniques can be used for remote sensing of bottom properties. However, in each specific case the experimental technique and analysis must focus on a specific wave type. In this paper, results are presented of experiments carried out in various shallow-water areas using an "L"-shaped array consisting of hydrophones in the water column and geophones on the seafloor. The total seismo-acoustic field measured with this array, including the pressure field in the water column and the head waves, diving waves, interface waves, and Love waves in the seafloor, is utilized to demonstrate the use of this approach as a remote sensing technique for seafloor parametrization.

Contributed Papers

3:30

3OC8. Reconstructing the seafloor properties from acoustic reflection data. John S. Papadakis, Michael I. Taroudakis, and Panagiotis J. Papadakis (Foundation for Res. and Technol.-Hellas, Inst. of Appl. and Comput. Math., P.O. Box 1527, 711 10 Iraklion, Crete, Greece)

The determination of the structure of the seafloor using measurements of the acoustic reflected power is an important inverse hydroacoustic problem with many applications in ocean geophysics. A realistic description of the seafloor includes many sedimentary layers over a harder substrate. Assuming that a rough estimation of the thicknesses of the various layers is available (by suitable subbottom profiling) the important thing is to recover the properties of the various layers, that is, the density, the sound velocity, and the attenuation coefficient. A simple scheme for the reconstruction of the seafloor is presented based on the assumption that reliable reflection coefficient measurements from a broadband acoustic source are available. The scheme is based on a systematic variation of the geoacoustic parameters of a background bottom model, so as the calculated reflection coefficient converges to the measured data. It is shown that data from various frequencies are necessary in order to obtain satisfactory results. [Work supported by EEC/MAST program.]

3:45

3OC9. Acoustic remote sensing of shallow ocean sediment characteristics. H. M. Chouhan and G. V. Anand (Dept. of Elec. Commun. Eng., Indian Inst. of Sci., Bangalore 560 012, India)

In this paper, a procedure is presented for estimating the acoustic parameters of shallow ocean sediments from the sound field measurements using a horizontal array. The sediment is modeled as a fluid with sound speed c at the water-sediment interface, a constant sound-speed gradient g , and a constant density ρ . Initially, the modal wave numbers k_m ($m = 1, 2, \dots, M$) are estimated using the generalized pencil-of-functions method. Next, an iterative procedure is used for estimating (c, g, ρ) . This procedure is based on the fact that the phase change ϕ_m of the modal waves at the water-sediment interface can be determined from the dispersion relation using the estimated values of k_m . The phase change $\hat{\phi}_m$ of the modal waves can also be computed for any assumed values of (c, g, ρ) . Estimates of (c, g, ρ) are obtained by adjusting their values so as to minimize the error $\epsilon = \sum_m (\phi_m - \hat{\phi}_m)^2$. Simulation results are presented to illustrate the method. [Work supported by DOE, Government of India.]

4:00–4:15

Break

Panel Discussion

PANEL MODERATOR: Robert D. Stoll

PANEL MEMBERS: Lester R. LeBlanc
George V. Frisk
J. F. Gettrust
John Diebold
Orest I. Diachok
Tuncay Akal

TUESDAY AFTERNOON, 30 APRIL 1991

CARROLL, 1:25 TO 5:30 P.M.

Session 3PA

Physical Acoustics: Sonochemistry and Acoustic Cavitation

Lawrence A. Crum, Chair

National Center for Physical Acoustics, Coliseum Drive, University, Mississippi 38677

Chair's Introduction—1:25

Invited Papers

1:30

3PA1. The past, present, and future of acoustic cavitation nucleation. Anthony A. Atchley (Physics Dept., Naval Postgraduate School, Monterey, CA 93943)

Those familiar with the field do not need to be reminded that acoustic cavitation is the cornerstone of sonoluminescence and sonochemistry. However, an equally important and often overlooked statement is that nucleation is the cornerstone of acoustic cavitation. So critical is the nature of the nucleus to the cavitation process that the acoustic cavitation threshold in water can be varied from a few atmospheres to a few hundred atmospheres just by altering the distribution and properties of the nuclei. (The acoustic cavitation threshold is the maximum acoustic pressure amplitude to which a liquid can be exposed before cavitation inception.) The fundamentals of acoustic cavitation nucleation will be presented in this paper, which is intended to be tutorial in nature. Topics include homogeneous and heterogeneous nucleation, stabilization mechanisms, models of heterogeneous cavitation nuclei, and the dependence of the cavitation threshold on the properties of the nucleus. [Work supported by ONR and the NPS Res. Program.]

1:55

3PA2. Gas temperatures in bubble oscillations. A. Prosperetti and V. Kamath (Dept. of Mech. Eng., Johns Hopkins Univ., Baltimore, MD 21218)

Sonoluminescence effects are critically dependent on the temperature field of the gas contained in oscillating bubbles. Polytropic models of the gas pressure-volume relationship are totally inadequate for an accurate determination of the temperature. A better model based on the conservation equations of continuum mechanics is described and its predictions illustrated with a number of examples. The influence of the computed temperature field on a model dissociation reaction is illustrated. [Work supported by NSF.]

3PA3. Experimental aspects of sonoluminescence. Ronald A. Roy and D. Felipe Gaitan (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

It is well known that acoustically driven gas bubbles can generate light through a process known as sonoluminescence. Although readily observed, this phenomenon has, until recently, been poorly understood. In the past, the existence of competing models (i.e., hot spots, electrical microdischarge, etc.) inspired several "definitive" experiments, many of which led to ambiguous or erroneous conclusions. Since then, improvements in our ability to detect photons as well as monitor and manipulate bubble oscillations have resulted in a significant enhancement of our understanding of the sonoluminescence process. This talk will begin with a brief review of the early theories and experiments, to be followed by a detailed description of recent experiments involving acoustically levitated, light-emitting bubbles. At issue is the timing of light output *vis á vis* the bubble collapse, the role played by the equilibrium size of the bubble and the acoustic pressure amplitude, the estimated collapse temperatures, and the contribution of various cyclic cavitation processes to the dynamics of single and multiple-bubble cavitation fields. [Work supported by ONR.]

2:45

3PA4. Shape oscillations of bubbles driven by modulated ultrasonic radiation pressure: Experiments. Philip L. Marston, Thomas J. Asaki (Dept. of Physics, Washington State Univ., Pullman, WA 99164-2814), and Eugene H. Trinh (Jet Propulsion Lab., Pasadena, CA 91109)

The shape of a bubble in water should change in response to the radiation pressure of an ultrasonic wave. Furthermore, modulation of the radiation pressure at the resonance frequencies for shape oscillations should facilitate the stable excitation of such modes [P. L. Marston, J. Acoust. Soc. Am. 67, 15-26 (1980)]. Recently this method of driving oscillations has been demonstrated by employing a novel ultrasonic levitator developed at Jet Propulsion Laboratory which traps bubbles in the size range of 1- to 5-mm diameter. The quadrupole mode was observed with resonance frequencies in the range from 400 to 40 Hz. Oscillations were detected with the unaided eye and with television and laser light scattering methods. These experiments suggest methods for investigating the nonlinear dynamics of bubbles, effects of surfactants, and the dynamics of bubbles in low gravity. Some comparisons with the dynamics of drops will be noted. [Research supported by ONR and NASA.]

3:10

3PA5. Sonoluminescence. Brad Barber, Ritva Löfstedt, and Seth Putterman (Dept. of Physics, Univ. of California, Los Angeles, CA 90024)

Sonoluminescence (SL) is an extraordinary phenomenon in that a standing wave sound field with an energy of 10^{-12} eV per atom can focus to such an extent that photons are emitted with energies that can be greater than 1 eV. The phenomenon has been known for over 50 years and the long held picture that it is due to acoustic cavitation has been demonstrated beyond a shadow of a doubt by the recent work of Gaitan and Crum [J. Acoust. Soc. Am. Suppl. 1 87, S141 (1990)]. They measured the dynamic radius, light emission, and sound field of a single bubble. Here, their pioneering work is extended to determine the details of the individual photon bursts and it is found that they last less than 2 ns and incude about a million photons. From a more general perspective the issue is raised of: Do cooperative optical processes affect the bursts and what are the limits of amplification that can be achieved via the self-focusing effects which lead to SL? Calculations suggest that our current acousto-optic conversion efficiency of 10^{-5} can be increased by at least 3 orders of magnitude. [Work supported by the Dept. of Energy, Office of Basic Energy Sci.]

3:35-3:45

Break

3:45

3PA6. The sonochemical hot spot. Kenneth S. Suslick (School of Chem. Sci., Univ. of Illinois at Urbana-Champaign, 505 S. Mathews Ave., Urbana, IL 61801)

The origin of "sonochemistry" is acoustic cavitation: the formation, expansion, and implosive collapse of bubbles in liquids irradiated with ultrasound. The compression of such bubbles generates intense local heating, which has been quantified recently from both chemical kinetic thermometry and from high-resolution sonoluminescence spectra. The temperatures reached during cavitation are ≈ 5000 K, but have an effective lifetime of only a few microseconds. Consistent with this, the sonoluminescence that accompanies sonochemistry closely resembles flame emission! The chemistry generated by these hot spots is different than

either ordinary thermal or photochemical processes and sonochemistry represents a fundamentally unique interaction of energy and matter. [For recent reviews see K. S. Suslick, *Sci. Am.* **260**, 80 (Feb. 1989) and *Science* **247**, 1439 (1990).] Recently, the use of ultrasound in liquid-powder slurries to enhance dramatically their chemical reactivity has been explored. For example, heterogeneous catalysis can be induced in normally nonreactive metals and the catalytic activity of Ni has been enhanced by 10^5 . Using a variety of surface science techniques, it was shown that ultrasound removed the passivating oxide coating normally found on Ni and other metal surfaces, thus increasing their activity. The origin of these effects comes from extremely high-speed interparticle collisions which occur during ultrasonic irradiation of liquid-solid slurries. Turbulent flow and shockwaves produced by acoustic cavitation can drive metal particles together at sufficiently high velocities to induce melting upon collision. A series of transition metal powders have been used to probe the maximum temperatures and speeds reached during such interparticle collisions. Metal particles that are irradiated in hydrocarbon liquids with ultrasound undergo collisions at roughly half the speed of sound and generate localized effective temperatures between 2600 °C and 3400 °C at the point of impact. [Work supported by NSF and the UIUC Materials Res. Lab.]

4:10

3PA7. Sonochemistry of volatile and nonvolatile solutes in aqueous solutions: A spin trapping study.

Peter Riesz (Radiation Oncology Branch, Natl. Cancer Inst., Natl. Inst. of Health, 9000 Rockville Pike, Bethesda, MD 20892)

The very high temperatures and pressures induced by acoustic cavitation in collapsing gas bubbles in aqueous solutions exposed to ultrasound lead to the thermal dissociation of water vapor into H atoms and OH radicals. Their formation has been confirmed by spin trapping for continuous wave and pulsed ultrasound. Sonochemical reactions occur in the gas-phase (pyrolysis reactions), in the gas-liquid interface, and in the bulk of the solution (radiation-chemistry reactions). The high-temperature gradients in the interfacial regions lead to pyrolysis products from nonvolatile solutes present at sufficiently high concentrations. The sonochemically generated radicals from carboxylic acids, amino acids, dipeptides, sugars, pyrimidine bases, nucleosides, and nucleotides were identified by spin trapping with a nonvolatile nitroso spin trap. At low concentrations of nonvolatile solutes, the spin trapped radicals produced by sonolysis are due to H atom and OH radical reactions. At higher concentrations of these nonvolatile solutes, sonolysis leads to the formation of additional reactions due to pyrolysis processes (typically methyl radicals). A preferred localization of nonvolatile surfactants at the gas-liquid interface was demonstrated. The volatile solutes methanol, ethanol, acetone, and acetonitrile were studied over the complete range of solvent composition. By the use of rare gases with different thermal conductivities, the contributions of individual reaction steps with widely different energies of activation can be evaluated.

4:35

3PA8. Industrial applications of high-intensity ultrasound. Raymond L. Hunicke (Lewis Corp., 102 Willenbrock Rd., Oxford, CT 06483)

With the increased laboratory investigations of sonochemistry, large-scale industrial applications can be expected in the near future. The ultrasonic manufacturing industry can provide large scale equipment with power ratings in the tens of kW and with flow rates of thousands of gallons per hour. For these scales, magnetostrictive transducers have strong relative advantages over piezoelectric transducers. Prior commercial applications of ultrasound to large-scale cleaning, aluminum soldering, and coal and ore beneficiation are discussed.

5:00-5:30

Bull Session

Session 3PP

Psychological and Physiological Acoustics: Session Honoring J. C. R. Licklider

Irwin Pollack, Chair

*Mental Health Research Institute, University of Michigan, Ann Arbor, Michigan 48109-0720**Invited Papers*

1:00

3PP1. J. C. R. Licklider, a remarkably versatile scientist. Irwin Pollack (Mental Health Res. Inst., Univ. of Michigan, Ann Arbor, MI 48109-0720)

An overview of the remarkably diverse scientific contributions of J. C. R. Licklider will be presented. Important contributions include: electrophysiology of the auditory cortex, design of speech communications systems, theories of hearing, human factors in command-control systems, the design of computers for human use, and the expedition of computer sciences as an academic discipline.

1:10

3PP2. J. C. R. Licklider as a psychoacoustician. Karl D. Kryter (SRI Intl., Menlo Park, CA 94025 and San Diego State Univ., San Diego, CA 92182)

Licklider's training and experience as a mathematician, physicist, and psychologist provided the basis for a unique variety of scientific endeavors in the fields of audition and perception. Some selected early contributions of Licklider to physiological and psychological acoustics will be briefly reviewed.

1:30

3PP3. J. C. R. Licklider, psychologist. George A. Miller (Dept. of Psychol., Princeton Univ., Princeton, NJ 08544-1010)

Personal recollections of working with Licklider at Harvard and MIT between 1943 and 1955, while he still considered himself to be a psychologist. At the Harvard Psycho-Acoustic Laboratory he conducted his definitive studies of the effects of amplitude modulation on the intelligibility of speech and proposed a theory of pitch perception. At MIT in the early 1950s, while continuing his interest in psychoacoustics, he also created a graduate program in experimental psychology and contributed to the creation of the Lincoln Laboratory.

1:50

3PP4. J. C. R. Licklider as computer scientist. John A. Swets (BBN Labs., Bolt Beranek and Newman Inc., 10 Moulton St., Cambridge, MA 02138)

Licklider was a national leader in the development of computer science. His theme was to make computers easier for humans to use, in the interests of aiding human cognition—what he termed “man-computer symbiosis.” Some of his seminal ideas, as developed mainly at BBN during the years 1958–1962, will be described and illustrated.

2:10

3PP5. J. C. R. Licklider and the case of the missing fundamental. Reinier Plomp (Dept. of Oto-rhino-laryngology, Free Univ. Hospital, P.O. Box 7057, 1007 MB Amsterdam, The Netherlands)

In 1955, Licklider visited The Netherlands and demonstrated by means of masking noise that the presence of the fundamental is not essential for hearing the low pitch of a complex tone. The “case of the missing fundamental” was a very hot issue in this country after Hoogland, in his 1953 doctoral dissertation, suggested that Schouten's opposite conclusion from experiments with signals without the fundamental, carried out in 1938–40, should be explained by nonlinear distortion in his equipment reintroducing the fundamental. Of course, Schouten did not accept this ill-based criticism. He considered his results as evidence that the low pitch is based on the periodicity of the combined harmonics not resolved by the ear,

his famous "residue" theory. With Licklider's demonstration, the classical controversy, going back to Seebeck and Ohm, seemed to be solved definitely in favor of Seebeck. "Periodicity pitch" theory, as elaborated by Licklider in his "duplex" (1951) and "triplex" (1955) theories, seemed to have succeeded "place pitch" theory, promoted by famous scientists such as Helmholtz and von Békésy, but for how long? The paper will review the significance of Licklider's contribution in the light of subsequent experimental evidence showing that frequency resolution does play an essential role in pitch perception.

2:30-2:45

Break

Contributed Papers

2:45

3PP6. Sharing place, period, and concepts with Licklider. I. J. Hirsh and P. G. Singh (Washington Univ. and Central Inst. for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

The nature of pitch as a dual attribute comprising both "place" and "period" aspects was dramatically illustrated by Licklider [J. Acoust. Soc. Am. **26**, 945 (1954)] in his famous masking demonstration. Listener confusions in judgments of pitch in subsequent experiments could be attributed to conflicting "place" and "period" information. Although the concept of pitch duality has been updated by pitch theorists, one aspect that has been consistently overlooked is the relation between "place" or "spectral" pitch and timbre. For example, the direction of a perceived pitch change is often confused with an ordinal aspect of timbre, like "brightness" and "sharpness." The connection between "place" pitch and timbre will be discussed and demonstrated with examples from the literature and from our own experiments. [Work supported by USAFOSR and NIH.]

3:00

3PP7. Effects of signal-to-noise ratio on the frequency discrimination of complex tones with overlapping and nonoverlapping harmonics. Brian C. J. Moore and Brian R. Glasberg (Dept. of Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB2 3EB, England)

Frequency difference limens for multiple-component complex tones (DLCs) were measured using an adaptive two-interval, two-alternative forced-choice task. The tones were presented either in quiet or in pink noise. The tones to be discriminated either had all harmonics in common or no common harmonics. DLCs for the tones with no common harmonics were generally, but not always, larger than those for complex tones with common harmonics. For the former, performance did not worsen "monotonically" with increasing noise level, but tended either to stay constant or improve at first, only worsening when the tones were almost completely masked by the noise. For the latter, performance tended to worsen "monotonically" with increasing noise level, although, again, large changes only occurred when the tones were almost completely masked by the noise. Even at the highest noise level used, DLCs for complex tones with no common harmonics were usually larger than DLCs for tones with common harmonics. The results suggest that the worse performance for tones with no common harmonics does not result from internal noise in the channels conveying information from the periphery to the mechanism determining residue pitch. Rather, spectral differences between tones with noncoincident harmonics appear to have a distracting effect that impairs pitch discrimination.

3:15

3PP8. Dichotic pitch discrimination. William A. Yost (Parham Hearing Inst., Loyola Univ., 6525 N. Sheridan Rd., Chicago, IL 60626)

Dichotic Huggins pitch was produced by generating broadband noise stimuli with narrow sections of the noise (bandwidths of 2, 4, 8, 16, 32, 64, and 128 Hz) interaurally phase shifted (by 90°, 135°, or 180°). The center frequencies of the narrow dichotic bands to which the interaural phase shifts were added were varied to change the value of the Huggins pitches. The just discriminable differences (obtained in a two-alternative, forced-choice adaptive procedure) in center frequency were determined for six listeners and at four (250, 400, 500, and 750 Hz) base center frequencies in an attempt to describe the pitch discrimination of dichotic Huggins pitches. For some listeners, at base center frequencies of 250 and 400 Hz and when the interaural phase shift used to produce the pitch was 180°, discrimination was almost as acute for these dichotic pitches ($\Delta f/f = 0.05\%$) as it was for diotically produced pitches. Pitch discrimination became more difficult as the center frequency increased to 500 and 750 Hz, the bandwidth of the interaurally altered band decreased from 128 to 2 Hz, or the interaural phase shift was reduced from 180° to 135° and 90°. The results will be discussed in terms of models of binaural processing and the use of interaural variables to segregate one sound image from other sound images. [Work supported by the NIDCD and the AFOSR.]

3:30

3PP9. Interaural correlation and the discrimination of one from two delayed sources. Irwin Pollack (Mental Health Res. Inst., Univ. of Michigan, Ann Arbor, MI 48109-0720)

In 1948, J. C. R. Licklider provided a powerful methodology for exploring the binaural system in terms of the cross correlations of signals presented to the two ears. The Licklider procedure is here extended to the discrimination between: (1) a binaural source with single ongoing interaural delay and (2) a combination of two binaural sources, each with its own delay, with the mean delay equal to the single delay. Accuracy of discrimination suffers at longer mean interaural delays. Accuracy of performance varies with the difference between the two delays of the combination. The direction of change is determined by the contrast in interaural correlation. The results are consistent with a binaural system where interaural correlation is evaluated at stations specific to the component interaural delay(s); where the strength of interaction between the component correlations decreases with the difference in the component interaural delays; and where the associated internal noise level increases with longer interaural delays (Jeffress). Licklider's methods and his correlational theory of binaural localization continue to provide a rich source of hypotheses and procedures for the analysis of binaural listening.

3PP10. Detection at the level of pitch. Ervin R. Hafter, Kourosh Saberi, and Eric R. Jensen (Dept. of Psych., Univ. of California, Berkeley, CA 94720)

Cues consisting of harmonics $f/2$ through $f/7$ have been shown to reduce frequency uncertainty and thus improve the detectability of a tonal signal at $f/1$ [E. R. Hafter and R. S. Schlauch, J. Acoust. Soc. Am. Suppl. 1 86, S112 (1989)]. However, a cue at $f/1$ did not affect the detectability of a signal consisting of only the higher harmonics. One reason for the asymmetry might be that the pure-tone timbre of a single frequency prevents its effectiveness as a cue for detection at the level of processing of pitch. Stimuli in the present study were three-tone sets, typically harmonics chosen at random from $f/2$ through $f/7$ from a sequence whose fundamental was itself chosen at random on each trial. For signals of constant level, best performance to worst was found with the following order: (a) cue = 3 harmonics (ex., $f/2, f/5, f/7$), signal = cue ($f/2, f/5, f/7$); (b) cue = 3 harmonics (ex., $f/3, f/6, f/7$), signal = other 3 harmonics ($f/2, f/4, f/5$); (c) no cue, signal = 3 harmonics; (d) no cue, signal = 3 randomly chosen inharmonic tones. These data are interpreted as showing that detectability can occur both at the level of the auditory filters and at the level of pitch.

3PP11. Is human hearing limited by Brownian motion? David M. Green and Huanping Dai (Psychoacoustics Lab., Psych. Dept., Univ. of Florida, Gainesville, FL 32611)

The normal human listener requires a signal energy to noise power density ratio (E/N_0) of 10 to reliably detect a 1000-Hz signal presented in a wide band noise. Since it does not seem likely that the process of detecting a sinusoidal signal depends on whether or not the noise is audible, a natural question is whether the absolute hearing sensitivity is limited by molecular motion. According to Watson, Franks, and Hood (1972), who were the first to control the signal duration (125 ms), the absolute threshold at 1000 Hz is about 9 dB SPL. The energy of this signal is approximately 10^{-16} W s/cm². We estimate the N_0 of Brownian motion over the same area to be about 1.7×10^{-20} W s/cm². Thus, the ratio of signal energy to noise power density at absolute threshold is about 5000. Our ability to hear weak auditory signals is impressive, but the fundamental limitation is not the thermal agitation of molecules of the air. [This research is supported by the AFOSR.]

TUESDAY AFTERNOON, 30 APRIL 1991

INTERNATIONAL D, 2:00 TO 4:05 P.M.

Session 3SA

Structural Acoustics and Vibration: Active Control of Structural Vibrations

Chris R. Fuller, Cochair

Department of Mechanical Engineering, Virginia Polytechnic Institute and State University, Blacksburg, Virginia 24061

Albert J. Tucker, Cochair

Code 1132, Office of Naval Research, 800 North Quincy Street, Arlington, Virginia 22217

Chair's Introduction—2:00

Contributed Papers

2:05

3SA1. A neural network adaptive controller for nonlinear processes. Randolph H. Cabell and Chris R. Fuller (Dept. of Mech. Eng., Virginia Polytech. Inst. and State Univ., Blacksburg, VA 24061)

Numerical simulations were used to study the use of an artificial neural network (ANN) as an adaptive controller for nonlinear applications. This implementation is similar to those using the adaptive LMS algorithm; however, the neural network can be used with nonlinear systems. The network contained two layers of internal processing units (hidden units), and the connection weights between units were calculated using the backpropagation training algorithm. The neural network was trained to control two types of nonlinearities: a square wave and a biharmonic system. The neural network controller achieved 15-dB attenuation when tested on the biharmonic system. In some cases the network required excessive time to converge to a solution, and the solution was sensitive to the initial configuration of the network. In general, the ANN shows potential for controlling nonlinear processes; however, there are a number of problems to be resolved before success-

ful applications are realized. [Work supported by ONR/DARPA and NASA.]

2:20

3SA2. Active vibration control of a flexible cylinder subjected to periodically shedding vortices. M. Kim (David Taylor Res. Ctr., Bethesda, MD 20084) and A. Baz (Catholic Univ. of America, Washington, DC 20064)

A mathematical model is developed to simulate the dynamics of a flexible cylinder subjected to vortex-induced excitation. The finite element method is used to describe the dynamic characteristics of the cylinder. The fluctuating lift forces induced on the cylinder are assumed to satisfy a modified Van der Pol equation and are determined by the wake-oscillator approach. The dynamic equations of the cylinder are coupled with the modified independent modal space control (MIMSC) method, which actively suppresses the dominant mode of the vibration. The performance of the active controller of the vortex-induced vibrations is obtained at various flow speeds and control weighting parameter. To verify the mathematical model, the computed results are com-

pared with experimental data obtained in a low-speed wind tunnel over a range of Reynolds numbers between 5500 and 7500. Good agreement is observed between the theory and experiments. It was found that the controller is effective in attenuating the transverse vibrations of the cylinder by approximately 40% for all the flow speeds considered.

2:35

3SA3. Excitation of thin beams using asymmetric piezoelectric actuators. Gary P. Gibbs and Chris R. Fuller (Dept. of Mech. Eng., Virginia Tech, Blacksburg, VA 24061)

In order to analyze the excitation of a complex beamlike structure comprised of arrays of piezoelectric actuators (that may not be arranged in symmetric pairs), it is necessary to determine the effect of a single asymmetric actuator. Using the superposition principle, the excitation of the entire structure can subsequently be analyzed. New analytical relations for the excitation of beams by a single piezoelectric element located on the surface of the beam (asymmetric excitation) were developed. A single asymmetric piezoelectric element when excited produces both flexural and extensional motion in a beam. The relations are used to study excitation of both flexural and extensional waves in beams using a pair of actuators as a function of: actuator length, relative voltage magnitude and phase, and frequency of excitation. The work presented here represents an important step because it is the analytical basis by which simultaneous active control of both flexural and extensional motion in flexible structures can be studied.

2:50

3SA4. Active vibration control using shear mode piezoceramic transducers. Chia-Chi Sung, Xiaoqi Bao, Vasundara V. Varadan, and Vijay K. Varadan (Dept. of Eng. Sci. and Mech., Res. Ctr. for the Eng. of Electron. and Acoust. Mater., Penn State Univ., 149 Hammond Bldg., University Park, PA 16802)

The " d_{15} " constant of the piezoceramics was utilized to design shear mode transducers. These shear mode transducers were verified to be pure shear stress/strain sensors/actuators both theoretically and numerically (ANSYS). Active torsional vibration control experiments were conducted on various phenolic and pyrex tubes by using the shear mode sensors/actuators. The free and controlled frequency responses as transfer functions were measured and compared to each other. The finite element mode-frequency analysis was performed to evaluate the structural characteristics for the tubes with and without transducers mounted. The dynamic coupling coefficients were derived for shear mode sensors/actuators based on the finite element mode shapes.

3:05

3SA5. Experiments on active control of acoustic radiation from discontinuities on infinite beams. Kenneth D. Frampton and C. R. Fuller (Dept. of Mech. Eng., Virginia Polytech. Inst. and State Univ., Blacksburg, VA 24061)

Two experiments were performed to study the active control of acoustic radiation from discontinuities on infinite beams. One experiment studied a semi-infinite/clamped beam and the other studied an infinite beam with a blocking mass. In each case the noise input was a subsonic bending wave incident on the discontinuity. Active control was implemented using LMS algorithm driving point force shakers or piezoceramic patches with far-field microphones as error sensors. In addition, an accelerometer array was employed to investigate changes in traveling and near-field waves under control conditions. Large attenuation in radiated sound at the error locations was noted while the beam vibrational amplitude often increased under control. Spatial Fourier analysis (wave number domain) of the beam vibration showed attenuation of supersonic wave numbers (responsible for acoustic radiation) while under control. Results for various frequencies and test configura-

tions will be discussed and mechanisms of control in terms of wave components considered. [Work supported by ONR/DARPA.]

3:20

3SA6. Active vibration control of a thin plate using piezoelectric sensors and actuators. S-Y. Hong, V. V. Varadan, and V. K. Varadan (Dept. of Eng. Sci. and Mech., Res. Ctr. for the Eng. of Electron. and Acoust. Mater., Penn State Univ., 149 Hammond Bldg., University Park, PA 16802)

Using piezoelectric sensors and actuators, the vibrating motions of a thin plate could be effectively controlled. Several pieces of Unimorph PZT (lead-titanate-zirconate) were optimally attached on the both sides of a clamped square plate. A series of active vibration control experiments has been tried with different types of control algorithms. Even though the system structure is stiff enough because of the clamped edges, effective vibration control could be implemented with the large electromechanical coupling constant of PZT and well-selected positions of sensors and actuators. Here, particular attention has been paid to several lower-frequency modes. To take advantage of the features of a digital system such as flexibility of the control program and increased logic capability, a microcomputer and a data acquisition board were used to obtain a digital controller.

3:35

3SA7. An experiment in state space control of a simply supported plate excited by steady disturbances. S. P. Rubenstein, W. R. Saunders, G. K. Ellis, and H. H. Robertshaw (Ctr. for Intelligent Mater. Systems and Structures, Mech. Eng. Dept., Virginia Polytech. Inst. and State Univ., Blacksburg, VA 24061)

Experimental results of an active vibration control experiment on a simply supported plate are presented. A linear quadratic Gaussian (LQG) regulator control design is implemented to reject steady disturbances of the plate's fundamental mode. A discussion of the control law is provided. Performance of a four-mode controller is demonstrated for harmonic, narrow-band, and broadband point-force disturbances. Typical reductions in the fundamental mode amplitude were up to 60 dB at the disturbance center frequencies. Results are presented in both time domain and frequency domain format. The digital controller was implemented on a transputer-based parallel processing system for enhanced computational performance. The initial sample speeds obtained with this system were up to eight times faster than the controller hardware previously available. [Work supported by DARPA/ONR.]

3:50

3SA8. Experiments on active isolation using PVDF distributed error sensors. Sylvie Lefebvre, Catherine Guigou, and Chris R. Fuller (Dept. of Mech. Eng., Virginia Polytech. Inst. and State Univ., Blacksburg, VA 24061)

The work described in this paper is an experimental appraisal of active techniques for the isolation of harmonic vibrations. In particular, the use of polyvinylidene fluoride (PVDF) strips as distributed error sensors on a receiving plate is studied and contrasted to the use of point sensors such as accelerometers. The experimental rig consists of an upper vibrating beam supported via natural rubber passive isolators on a receiving plate with two clamped boundaries. The control approach used is a two-channel adaptive LMS algorithm implemented on a Texas Instrument TMS320C25 DSP chip. The control forces are applied through electromagnetic devices, in parallel with the passive isolators. In general, the results show that the PVDF sensors improve the control performance in terms of global reduction of the response of the receiving structure, by limiting control spillover. The behavior of the system

in terms of attenuation is demonstrated to be strongly dependent upon the forcing frequency and the location and nature of the error sensors. For example, the use of the accelerometers located close to the control

actuators, while providing high local reduction, can lead to an increase of the global response. [Work supported by NASA Langley and DARPA/ONR.]

TUESDAY AFTERNOON, 30 APRIL 1991

INTERNATIONAL B, 1:00 TO 4:15 P.M.

Session 3SP

Speech Communication: Speech Processing

Astrid Schmidt-Nielsen, Chair

Code 5532, Naval Research Laboratory, Washington, DC 20375-5000

Contributed Papers

1:00

3SP1. Profiling vectors for speaker identification. Harry Hollien and Ming Jiang (Linguistics and Inst. for Adv. Study of the Commun. Processes, Univ. of Florida, Gainesville, FL 32611)

When speaker verification is the issue of interest, it is possible to focus on signal analysis irrespective of the speech related features it contains. Such approaches are appropriate in this case because system distortions are minimal, noise is low, talkers are cooperative, and very sophisticated equipment is available. Not so for speaker identification. Here extensive channel and speaker distortions (including noise) can be expected; speech is noncontemporary and speakers usually uncooperative. Hence, the signal is so distorted or masked, the usual processing techniques cannot be expected to be very useful. The approach to speaker identification demonstrated in this paper is threefold. First, it is assumed that the signal contains speech features that are robust (i.e., resistant to noise and distortion) and unique to the talker. These idiosyncracies are based on speaker's anatomy, physiology, and habitual communicative patterns. Second, it is postulated that, while there may be no single attribute within a person's speech/voice that would permit them to be differentiated from all other speakers under any set of conditions, the simultaneous use of a large series of feature analyses may permit identification. Finally, it has become possible to reduce bias among the vectors by the normalization of data. In turn, this approach leads to a very effective two-dimensional profile wherein the unknown speaker must first be identified and then comparisons made to known talkers. A system of this type has been structured and tested; it is based on four natural speech vectors, each containing 20-40 parameters. Data regarding this general approach and these vectors have been reported previously. This presentation will focus on the effects (on efficient speaker identification in the field) of normalizing the vector data and reducing it to a two-dimensional profile.

1:15

3SP2. Vowel formant tracking for speaker identification. Ming Jiang and Min Shi (Inst. for Adv. Study of the Commun. Proc., Univ. of Florida, Gainesville, FL 32611)

The first two or three spectral peaks, or formants, are crucial in determining the vowel quality. In turn, accurate determination vowel formants quality is important to effective speaker identification task. A vowel formant tracking vector (VFT) was developed for the speaker identification (SAUSI) profile. Specifically, the speech spectrum is ob-

tained frame-by-frame by using an LPC algorithm with the first three formant frequencies for each frame calculated. The underlying assumption was that the vowels will exhibit a contiguous formant frequency transition from frame-to-frame and, hence, can be separated from consonants for the cited formant measurements. In order to carry out this task, the frequency range 0-5000 Hz is divided into 34 semitone bins and three histograms are obtained for first three vowel formants. In turn, these histograms provide an estimation of general quality of the vowels spoken by each speaker being evaluated. The result is that the interspeaker differences are large enough to permit identification of the target speaker while the intraspeaker differences are fairly small even for text independent speech. The algorithm utilized will be presented as will data demonstrating that this VFT vector is robust enough to effectively perform the speaker identification task.

1:30

3SP3. SAMREC0: A C30-based reference connected-word recognizer for the evaluation of speech databases. F. Capman and G. Chollet (C.N.R.S. URA 820, Télécom Paris, Dépt. Signal, 46 rue Barrault, 75634 Paris Cédex 13, France)

One of the objectives of the ESPRIT-SAM project is the elaboration of speech databases for the evaluation of recognizers. In this framework, a reference system [G. Chollet and C. Cagnoulet, "On the evaluation of speech databases using a reference system," ICASSP, 1982], based on dynamic programming algorithm, was modified to accept connected words [G. Chollet and C. Montacie, "Evaluating speech recognizers and databases," NATO-ASI, 1988]. This software, which is called SAMREC0 by the SAM speech input assessment group, is now implemented using a T.I. TMS320C30-based PC-board, so that it can be used efficiently on the SAM PC-AT workstation. Some results will be presented on the evaluation of the first SAM database EUROM0. This database was recorded in quiet conditions and very few classification errors are observed. Work is under development to simulate noisy conditions using the same database, in order that the limits of the reference or other systems could be measured.

1:45

3SP4. Feature detection using a connectionist network. Gary Bradshaw (Dept. of Psych., Univ. of Colorado, Boulder, CO 80309) and Alan Bell (Univ. of Colorado, Boulder, CO 80309)

A feedforward connectionist network trained by backpropagation was used to detect 15 speech features. The network was trained over 240 sentences (40 men and 40 women), and tested over 200 sentences (10 men and 10 women), all part of the MIT Ice Cream database. Network input consisted of a smoothed spectral vector at 15-ms-intervals, plus two coefficients of amplitude and spectral change. The network achieves a signal detection discrimination level (a -prime) of 0.87 compared to a level of 0.76 for a ten-nearest-neighbor system. Almost identical training and test performances indicates excellent generalization to new speakers and text. Processing costs are mainly signal processing and network training; detection itself can be done in real time. Performance is much better for broad features like sonorance, which occur frequently, than for infrequent features like sibilance, partly because of their low frequency and partly because of other characteristics. [Work supported by USWest.]

2:00

3SP5. Neural networks in articulatory speech analysis/synthesis. M. G. Rahim, W. B. Kleijn, and J. Schroeter (AT&T Bell Laboratories, Murray Hill, NJ 07974)

A major difficulty in articulatory analysis/synthesis is the estimation of vocal-tract parameters from input speech. The use of neural networks to extract these parameters is more attractive than codebook look-up due to the lower computational complexity. For example, a multilayer perceptron (MLP) with two hidden layers, trained and evaluated on a small data set was shown to perform a reasonable mapping of acoustic-to-geometric parameters. Increasing the training data, however, revealed ambiguity in the mapping that could not be resolved by a single network. This paper addresses the problem using an assembly of MLP's, each designated to a specific region in the articulatory space. Training data were generated by randomly sampling the parameters of an articulatory model of the vocal system. The resultant vocal-tract shapes were clustered into 128 regions, and an MLP with one hidden layer was

assigned to each of these regions for mapping 18 cepstral coefficients into ten tract areas, and a nasalization parameter. Networks were selected by dynamic programming, and were used to control a time-domain articulatory synthesizer. After training, significant perceptual and objective improvements were achieved relative to using a single MLP. Comparable performance to codebook look-up with dynamic programming was obtained. This model, however, requires only 4% of the storage needed for the codebook, and performs the mapping faster by a factor of 20.

2:15

3SP6. Automatic speech recognition based on property detectors. T. V. Ananthapadmanabha and H. N. Jayasimha (Voice and Speech Systems, 669, I Floor, 20th Cross, II Block, Rajajinagar, Bangalore 560 010, India)

Speaker-independent, large-vocabulary, continuous speech recognition by a machine is a challenging problem for which over a decade of research has been made without significant progress. In the existing systems, the *same* acoustic feature vector (LPC, cepstrum, filter bank, etc.) is used for all speech sounds and they heavily depend on contextual information for their success. This paper presents some results based on a radically different approach called "property detectors." The approach of property detectors is well known in visual perception where it has been demonstrated that specialized detectors exist on the retina that trigger only for vertical, horizontal, or inclined lines. It has only been speculated that such specialized detectors could exist for speech. Recently, acoustic properties have been discovered that uniquely characterize some phonemes like /a/, /i/, /u/, /e/, /o/, and /s/. A limited-vocabulary, speaker-independent airline schedule announcement system was developed. This system was tested in a noisy hall with a large number of speakers, including female speakers, with different linguistic backgrounds. The system, though is in its early stage, gave a performance of about 85% accuracy. The approach based on property detectors appears promising.

2:30-2:45

Break

2:45

3SP7. Synthesis of manner and voicing continua based on speech production models. Corine Bickley, Kenneth N. Stevens (Res. Lab. of Electron., MIT, Cambridge, MA 02139), and Rolf Carlson (MIT, Cambridge, MA 02139)

The goal of this project is to create natural-sounding synthetic consonant-vowel syllables for presentation to aphasic patients and normal controls in studies of perception of speech sounds and lexical access. Of particular interest are the manner distinctions that appear to form the basis for the processing of other phonetic dimensions by human listeners. Continua of syllabic-nonsyllabic, sonorant-obstruent, continuant-noncontinuant, and voiced-voiceless sounds were constructed using the KLSYN88 synthesizer. The endpoint stimuli were synthesized based on theoretical models of glottal and turbulence noise sources and vocal-tract filtering, with some refinements to match the characteristics of a particular speaker. Intermediate stimuli were created to form continua that represent incremental changes in the synthesizer parameters. For all stimuli, the values of synthesis parameters modeled utterances that could be produced by a human talker. Identification functions for these continua for normal listeners showed relatively sharp boundaries between phonetic categories. The acoustic char-

acteristics of the stimuli in the vicinity of the boundaries were examined to determine the pattern of acoustic attributes responsible for the abrupt change in identification, such as rise times of amplitudes, rates of change of formants, and relative amplitudes of noise and glottal excitations. [Work supported in part by NIH grants DC00776 and DC00075.]

3:00

3SP8. Considerations on speaking style and speaker variability in speech synthesis. Lennart Nord and Björn Granström (Dept. of Speech Commun. & Music Acoust., Royal Inst. Tech., Box 70014, S-10044 Stockholm, Sweden)

In the exploration of speaking style and speaker variability, a multispeaker database and a speech production model is used. The structure of the database, which includes professional as well as untrained speakers, makes it possible to extract relevant information by simple search procedures. In perceptual studies both $F0$ and duration has had an indisputable effect on prosodics but the role of intensity and of segmental variation has been less clear. This has resulted in an emphasis on the former attributes in current speech synthesis schemes. Intensity has a

dynamic aspect, discriminating emphasized and reduced stretches of speech. A more global aspect of intensity must be controlled when an attempt is made to model different speaking styles. Specifically, attempts have been made to model the continuum from soft to loud speech. Systematic variation in speech synthesis has been used as a tool to explore possible speaker dimensions, among them reduced and over-articulated speech. Listening experiments have been carried out with the aim to investigate whether it is possible to describe synthesis samples according to different attitudinal and emotional dimensions.

3:15

3SP9. Improvement of synthetic speech quality through syntactic information. Tohru Shimizu, Seiichi Yamamoto, Norio Higuchi, and Hisashi Kawai (KDD R&D Labs., 2-1-15 Ohara, Kamifukuoka-shi, Saitama 356, Japan)

Many words in Japanese have identical written expression but different pronunciation. Natural synthetic speech therefore requires selection of the correct pronunciation for words and optimized prosodic features, including accent position and level, sentence intonation and length of pause, through the use of syntactic features. This paper describes (1) a new method of determining phrase accent level, based on accentual phrase boundary location and compound word structure, and (2) a newly proposed syntactic class of phrase boundaries. The results of the automatic determination of pronunciation, and opinion tests of intelligibility and naturalness are also described. About 10 000 words are assigned to syntactic and semantic features to determine correct pronunciation, representing about 20% of the total vocabulary. Pronunciation of 99% of the words in a Japanese economic daily were correct, and naturalness of the synthetic speech was 1.1 grades higher under the five-grade opinion test.

3:30

3SP10. Source parameters for the fricative consonants /s,ʃ,ç,x/. Christine H. Shadle (Dept. of Electron. and Comput. Sci., Univ. of Southampton, Southampton SO9 5NH, U.K.)

A series of experiments with mechanical models of fricative consonant articulatory configurations have been conducted to determine where in the tract the turbulence noise is generated and the spectral characteristics of that noise. The latest models, based on a combination of x ray, EPG, and photographic data, have the correct midsagittal profile and area function, and thus have the most realistic shape of model work to date. Data obtained from /s,ʃ/ substantiate earlier results based on a different subject [C. H. Shadle, *J. Acoust. Soc. Am. Suppl.* 1 84, S34 (1988); C. H. Shadle, in *Speech Production and Speech Modelling*, Proc. of NATO-ASI, edited by W. Hardcastle and A. Marchal (Kluwer Academic, Amsterdam, 1990), pp. 127-219] and results from extremely idealized models [C. H. Shadle, Proc. 12th ICA, paper A3-4, Toronto (1986)]. Comparisons across a range of flow rates, with and without sublingual cavity, between measured source and far-field spectra, and between speech and model data for /s,ʃ,c,x/ lead to source parameters, a distinction between two source types, and to the conclusion that the three-dimensional shape of the tract is crucial in determining source parameters: these parameters can be used in a model based

on one-dimensional sound propagation. Three-way comparisons between far-field sound measured (1) for the models and (2) for actual utterances, and (3) far-field sound predicted from measured source parameters used in a model based on one-dimensional sound propagation, will be shown. [Work supported by SERC.]

3:45

3SP11. Reliable glottal-closure-instant (GCI) estimation from short analysis frames. Krishna S. Nathan^{a1} and Harvey F. Silverman (LEMS, Div. Eng., Brown Univ., Providence, RI 02912)

It is well known that the first formant is maximally excited at the instant of glottal closure. Therefore, it is natural to utilize the energy in a band containing the first formant as a cue to the GCI. In practice, however, the actual GCI lies a few samples prior to where this energy signal attains a local maximum. Moreover, such an estimate makes no use of any period information regarding the GCI's. Consequently, secondary excitations within a period can lead to spurious GCI's. It is therefore proposed to augment the information contained in the first formant with the linear prediction error. Although, prediction error has been widely used for pitch determination, it is not sufficient to locate the GCI reliably because of ambiguities arising from multiple peaks, especially for vowels like /u/ (as in *foot*). Interestingly, these experiments have shown that secondary excitations tend to result in peaks in the residual error signal at locations different from those in the formant energy signal. Furthermore, in the absence of spurious excitation, the residual error can contain valuable independent period information. Therefore, the product of these two signals yields accurate GCI estimates. Such an algorithm has been tested on all vowels in a variety of environments and has been found to be very robust. Analysis frames as short as 5-10 ms have been used. ^{a1}Current address: IBM Research, T. J. Watson Research Center, Yorktown Heights, NY 10598.

4:00

3SP12. Isolation and characterization of microevents in speech. David A. Berry and William J. Strong (Dept. Physics and Astron., Brigham Young Univ., Provo, UT 84602)

An event-synchronous technique has been designed in an attempt to optimize time and frequency resolution in speech analysis. The technique isolates "microevents" in the speech waveform and then analyzes them, thus differing from commonly used asynchronous methods that employ a fixed frame length stepped forward in constant time increments. A microevent (ME) is associated with a "packet of energy" in the waveform and is initiated by some underlying input or fluctuation of energy. There are four basic types of MEs: (1) a voiced ME is initiated by a pitch pulse; (2) a plosive ME is initiated by a plosive burst; (3) a noise ME is initiated by a positive fluctuation in energy; and (4) a mixture ME. An ME is terminated at the initiation of the next ME or when the energy of the speech signal falls below the background level. ME durations are constrained to lie within a range of 2-20 ms. The current algorithm, developed and tested with portions of the 1988 DARPA TIMIT acoustic-phonetic continuous speech database, isolates over 95% of the MEs correctly. Once isolated, MEs are characterized by their one-third octave spectra. Results will be illustrated with various examples.

Session 3UW

Underwater Acoustics: Propagation

Evan K. Westwood, Chair

Applied Research Laboratories, University of Texas, Austin, Texas 78713

Chair's Introduction—1:25

Contributed Papers

1:30

3UW1. A modified perturbation approach for calculating the vertical wave function required in normal mode theory. Robert Zingarelli and M. F. Werby (NOARL, Theoret. Acoust. Code 221, Stennis Space Center, MS 39529)

Perturbation approaches are useful when one wishes to solve a problem that does not differ significantly from a known solution. The conventional approach to derive perturbation methods is to begin with a solvable solution χ , and then to expand the desired solution in terms of first- and higher-order contributions of the perturbing component—with a small coefficient ϵ —which differs from the exact solution only by the perturbing component. Then first-order perturbation theory is obtained by dropping higher powers of ϵ . In quantum physics this method is referred to as Fermi's second golden rule. This method can often fail as will be demonstrated and it can lead to erroneous results. This work reports on a new and refined perturbation method based on completeness considerations in which the exact expansion solution in matrix form is set up. Gauss-Seidel iteration is then employed and shows that the first iteration leads to an improved first-order perturbation theory, the second iteration leads to an improved second-order perturbation theory, etc. The method is illustrated with a few examples in which the old and new methods are compared with the exact solution for the case of vertical wave functions for variable speed profiles perturbed about an isovelocity case.

1:45

3UW2. On calculation of the acoustic field in a water layer with variable depth placed on a liquid half-space. Valery B. Galanenko (Dept. of Acoust., Kiev Polytech. Inst., Prospect Pobedy 39, Kiev, 252056, USSR)

The wave problem for the variable depth Pekeris waveguide with special cross sections is discussed by means of the cross-section method. The wave field is expanded as the sum of a series and integrated with a regular kernel or by summing a series of mode components only. Both are equivalent to the usual expansion as sums of coupled modes and waves of continuous spectra. A system of coupled differential and integral equations or an equivalent set of differential equations only is obtained. Some numerical results are presented and it is shown that mode reflection coefficients for a near-coast wedge with a typical bottom are less than 0.03 if the incident angle $\theta = 0.03$ and they increase to 0.8 when $\theta = 60$.

2:00

3UW3. A high-order adaptive integration method for wave propagation in range-independent fluid-solid media. Sven Ivansson and Ilkka Karasalo (Dept. of Hydroacoust. and Seismol., Swedish Defence Res. Est., S-17290 Sundbyberg, Sweden)

A high-order, adaptive method is described for computing the wave field in a laterally homogeneous fluid-solid medium by Henkel-transform integration. A technique for numerical quadrature is used, where trapezoidal or Filon sums obtained with several step sizes are combined by polynomial or Bulirsch-Stoer rational interpolation to increase the order of convergence and to obtain error estimates. This technique is combined with adaptive interval halving, maintaining a hierarchy of subintervals, meshes, and function values in a stack to eliminate duplicate function evaluations. Computational results from an underwater acoustics application are presented, showing impressive gains in efficiency and accuracy in comparison to traditional nonadaptive methods. Wave-field computations with tight accuracy demands are of interest, specifically in the context of benchmark calculations and program verification. With the method presented, about five additional correct digits are obtained by increasing the computational work by a factor of two. Of particular interest is the observation that the Filon technique seems really attractive only in conjunction with the adaptive approach.

2:15

3UW4. Simulating acoustic fields with a lattice gas model for waves. S. K. Numrich, R. A. Krutar, R. K. Squier (U.S. Naval Res. Lab., Washington, DC 20375-5000), and J. Pearson (Los Alamos Natl. Lab., Los Alamos, NM 87545)

Lattice gas models have been useful in simulating complicated field behavior and are particularly well suited to take advantage of massively parallel computer architectures. As an alternative to the traditional computational methods used in acoustics, a lattice gas model based on the wave equation has been implemented on a Connection Machine and used to simulate acoustic phenomena. The model has been successful in simulating independently driven point sources, functioning singly, as pairs and arranged in both traditional line arrays and a variety of alternative configurations. The implementation of the model can also handle rigid and pressure release boundaries as well as changes in sound speed. Both scattering and propagation are directly computed in the same model. Once a phenomenon, a boundary condition or sound speed change, has been correctly modeled, it can be placed at any point or in any area anywhere in the field. The model then produces the time evolution of the pressure field everywhere in the modeled region. The results are shown as video tapes of that time evolution or snapshots of the entire field at some instant of time. [Work supported by ONT.]

3UW5. High resolution and efficient oceanographic and acoustic modeling of propagation through mesoscale oceanic features. L. S. Gardiner, B. R. Frederick, T. A. Chmielewski, P. Bilazarian, and W. F. Mitchell (General Electric Co., Adv. Technol. Lab., ATL Bldg., Moorestown Corp. Ctr., Moorestown, NJ 08057)

A procedure is presented for combining the output of advanced multilevel "primitive equation" dynamic oceanographic forecast models with high fidelity underwater acoustic propagation models. This procedure is appropriate for the development of realistic predictions of acoustic transmissions through complicated mesoscale features, such as currents, rings, eddies, and fronts. Examples of oceanographic predictions

using the Princeton dynamic ocean model developed by Mellor and Blumberg are displayed graphically for a dataset of the Gulf of Mexico. Results pertinent to data interpolation, the identification of mesoscale oceanic features, and 3-D visualization are presented. Ocean data are converted to sound-speed profiles for this region and are interfaced to a range-dependent parabolic equation (PE) propagation loss model. As a low-frequency acoustic source is moved successively from shallow to deep water in the Gulf of Mexico, results are given that illustrate large variations in propagation loss to a receiver located in deep water. The relative contributions of bathymetry and mesoscale oceanic features to these variations are analyzed. Finally, a scheme for speeding up PE calculations by utilizing the massively parallel computational capabilities of the Connection Machine at GE/ATL is discussed.

2:45-3:00

Break

3:00

3UW6. Derivation of one-way wave equations for vector wave propagation problems. Louis Fishman (Dept. of Math. and Comp. Sci., Colorado School of Mines, Golden, CO 80401)

The recent application of pseudodifferential and Fourier integral operators and functional integral methods to the factored scalar Helmholtz equation has resulted in extended parabolic wave theories, corresponding path integral solutions, and a numerical marching algorithm for a variety of acoustic wave propagation problems. These known techniques are applied here to vector wave equations (Maxwell's and elasticity), resulting in new first-order Weyl pseudodifferential equations, which are recognized as exact one-way wave equations for transversely inhomogeneous environments. Perturbation treatments of the appropriate Weyl composition equations for the operator symbol matrix yield high-frequency and other asymptotic wave theories. Unlike the scalar Helmholtz equation case, the one-way vector equations (and a scalar analog provided by the Klein-Gordon equation of relativistic physics) require the solution of generalized quadratic operator equations. While these operator solutions do not have a simple formal representation as in the straightforward (acoustic) square root case, they are conveniently constructed in the Weyl pseudodifferential operator calculus. This is an exact formulation at the level of the wave field—no special symmetry and/or far-field assumptions are made. [Work supported by NSF, AFOSR, ONR.]

3:15

3UW7. Generating eigenray tubes from two solutions of the wave equation. James B. Bowlin (Woods Hole Oceanographic Inst., Dept. of Appl. Ocean Phys. and Eng., Woods Hole, MA 02543)

A method is presented for calculating the paths taken by sound between a source and receiver. These paths, which are called eigenray tubes, are obtained from two solutions of the wave equation at finite frequency, one propagated from the source and the other propagated from the receiver. The results are not restricted to the high frequency limit as is the case with classical ray traces. This generalization of classical ray tracing could be an important new tool in acoustic tomography.

3:30

3UW8. A complex ray method using asymptotic expansions applied to the penetrable wedge. Grant B. Deane (Marine Phys. Lab., Scripps Inst. of Oceanography, La Jolla, CA 92093)

The acoustic field in a sloping, isovelocity waveguide (the penetrable wedge) is found by extension and application of a complex ray model. The pressure is computed as a sum of ray fields, each of which is expressed as an asymptotic expansion of a plane wave integral evaluated using the method of steepest descent. A geometrical interpretation of the method is given which should be possible to extend to three-dimensional environments.

3:45

3UW9. Magnus expansion and split-step PE error analysis. F. J. Ryan (Ocean and Atmos. Sci. Div., Code 541, Naval Ocean Systems Ctr., San Diego, CA 92152-5000)

A practical problem involved with implementing parabolic wave equation (PE) computer codes is the optimal selection of depth mesh and range step-size. Efficiency requires that a varying range step-size be used. An objective method for automatic range step-size selection in split-step PE codes can be derived by an error budget analysis based upon the Magnus expansion of an exponential operator. The method will be described and examples shown for PE propagation in range-dependent environments.

4:00

3UW10. Stability analysis of a higher-order time-domain paraxial approximation. B. J. Orchard, W. L. Siegmann, M. J. Jacobson, G. J. Habetler (Dept. Math. Sci., R.P.I., Troy, NY 12180-3590), and Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

The global and local stability of an n th-order time-domain paraxial approximation (TDPA _{n}) to an acoustic wave equation [M. D. Collins, J. Acoust. Soc. Am. **86**, 1097-1102 (1989)] is resolved. An operator splitting technique and the Trotter product formula are used to determine the conditional (i.e., parameter dependent) stability of the complete paraxial operator from the stability of the individual diffractive, refractive, and dissipative components. The local stability of a finite difference implementation of the TDPA _{n} is analyzed using Von Neu-

mann and matrix methods. The Von Neumann analysis, which provides a necessary (but not sufficient) stability condition, suggests that the algorithm is unconditionally stable. The matrix method provides two requirements. A necessary condition for stability on the spectral radius of the amplification matrix W is unconditionally satisfied. However, a sufficient condition on the norm $\|W\|_2$ is only conditionally satisfied. Consequently, the parameter and step-size values required for use of the algorithm are determined. These results are illustrated with numerical examples, and implications for use of the TDPA_n are described. The TDPA_n marches the solution in range. A similar stability analysis shows that an analogous algorithm that marches the solution in time is unstable. [Work supported by ONR.]

4:15

3UW11. Predictability of acoustic intensity in shallow water at low frequencies using parabolic approximations. R. J. Cederberg, W. L. Siegmann, M. J. Jacobson (R.P.I., Troy, NY 12180-3590), and W. M. Carey (Naval Underwater Syst. Ctr., New London, CT 06320)

The accuracy of predictions for acoustic intensity in shallow water from parabolic approximation models are carefully examined. First, accuracy comparisons are made between analytic solutions to the Helmholtz equation and parabolic equations for propagation in a Pekeris waveguide. The parameter dependencies of the interference patterns for frequencies below 100 Hz are described. Calculations from the IFD implementation of parabolic equations are also compared with the analytic solutions. Agreement between the analytical and numerical results is very good, providing that appropriate initial conditions are employed. Sensitivities to input parameters are examined by observing the variations in the predicted fields and intensities as the parameters are changed. The parameters include those specifying the depth and range properties of the geoacoustic bottom, sound-speed profile, and spatial features of the waveguide. Magnitudes of parameter variations are se-

lected consistent with a recent Hudson Canyon experiment. Comparisons of the experimental results with model predictions are discussed. One conclusion is that input parameter uncertainties, even though relatively small for this experiment, can in some cases significantly affect the predictions. [Work supported by ONR.]

4:30

3UW12. Perth, Bermuda (1962): A shot heard round the world according to adiabatic mode theory. K. Heaney,^{a)} W. A. Kuperman, and B. E. McDonald (Naval Res. Lab., Washington, DC 20375)

A 1962 experiment is examined in which sound from an underwater explosion near Perth, Australia, was detected near Bermuda. A recent attempt [Munk *et al.*, *J. Phys. Ocean.* **18**, 1876 (1988)] to calculate propagation paths for this event included rotational flattening of the Earth and horizontal refraction due to the mean equatorward temperature gradient. That calculation left Bermuda in a shadow zone. The current work invokes adiabatic mode theory to include refraction due to horizontal variations in the vertical mode structure. The results include separate horizontal rays for each of the first few vertical modes, using an archival data set of 230 ocean sound profiles to generate the modes numerically. Where appropriate, interaction with bathymetry was included (scattering and bottom losses). This solution possesses two eigenray groups: group A passes just south of the Cape of Good Hope, at which point group B is almost 1000 km to the south. Intermediate rays are blocked by islands. Group A proceeds unimpeded to Bermuda for a total time-of-flight of 13 360 s, while group B interacts slightly with bathymetry off Brazil, arriving 37 to 45 s behind group A, and suffering 8 to 10 dB more bottom attenuation. These numbers are within experimental uncertainties (main arrival at $13\,364 \pm 5$ s; pulse train half-width ~ 15 s; second arrival $\sim 30 \pm 5$ s later). [Work supported by ONR.] ^{a)}Permanent address: Planning Systems Inc., McLean, VA 22101.

Session 4AA

Architectural Acoustics: Newer Measurement Procedures in Auditoria III

Steven M. Brown, Chair

Steelcase, Inc., CD-5E-16, P. O. Box 1967, Grand Rapids, Michigan 49546

Invited Papers

9:00

4AA1. RAMSoft II: A computer-based room acoustics measurement system. R. E. Halliwell and J. S. Bradley (Inst. for Res. in Construction, Natl. Res. Council of Canada, Ottawa, Ontario K1A 0R6, Canada)

A computer-based measurement system has been developed to determine a number of the newer measures used in the acoustical evaluation of rooms. Although the output is similar to the earlier RAMSoft system from which it has evolved, this system is based on a quite different measurement concept. This is a two-microphone system using an MLS sequence as the excitation signal. Fast Hadamard transforms on the two signals provide both omnidirectional and figure of eight impulse responses for each measurement position. From these impulse responses 14 acoustical measures, including lateral fractions, are calculated. The reasons for the departure from RAMSoft will be discussed and measurements made with both this system and with the RAMSoft measurement system will be compared.

9:30

4AA2. All-scale model measurements: The MIDAS system. Xavier Meynial (Lab. d'Acoust., URA 1101, Université du Maine, 72017 Le Mans, France), George Dodd (Univ. of Auckland, Auckland, New Zealand), Jean-Dominique Polack (Univ. du Maine, 72017 Le Mans, France), and A. Harold Marshall (Univ. of Auckland, Auckland, New Zealand)

Acoustical measurements on models offer a time and cost efficient method for determining the acoustical properties of enclosures. In addition the use of models facilitates understanding as a result of the visible 3-D form of the models and the fact that the process can be interactive. Cooperation between Auckland and Le Mans Universities has resulted in the development of a computer-based measuring system for all-scale acoustical measurements. This system, intended for commercial availability, features a numerical compensation technique for air absorption, thus avoiding the need for air-drying or air-replacement and therefore allowing free access to models during testing. The present Macintosh-based system uses an FFT approach to compute the full range of accepted auditorium indices and other building acoustics measures (e.g., TL and sound power) using a range of possible sound sources both impulsive and continuous. It has been used successfully as a design aid in consultancy work on concert halls, for calibrating measurement chambers, and as a research and teaching tool. Results from specific applications and ideas for new uses will be presented.

10:00

4AA3. Binaural measurement analysis of performing arts spaces. Wade R. Bray (Sonic Perceptions, Inc., 114A Washington St., Norwalk, CT 06854)

Several performing arts halls are evaluated with a measurement-compatible calibrated dummy-head system and a digital binaural acoustic analyzer. Binaural impulse responses are taken and edited in the time domain, dissonant harmonic structures and their resulting beat-frequency effects in the range below 100 Hz are measured in one hall having a high perceived level of low-frequency noise not in keeping with the NC measurement in the same space. Immediate *A/B* comparisons for listening juries are made of symphonic sound with different stage conditions, as heard in different seats in the house, and compared using three-axis spectrographic measurement. Panel absorptions, a cavity resonance, loudspeaker tonal colorations, and other effects perceived by human hearing are measured and corrected.

10:30

4AA4. Subjective and objective evaluations of classrooms and lecture halls. Wei-hwa Chiang, Gary W. Siebein, and Richard P. Cervone (Dept. of Arch., 231 ARCH, Univ. of Florida, Gainesville, FL 32611)

Speech intelligibility tests were given to groups of college students in ten classrooms and lecture halls of various sizes on the University of Florida campus. A series of acoustical measurements including reverberation time, early reverberation time, loudness, early to late temporal energy ratios, lateral energy fractions, interaural cross correlation, and speech transmission index were made at multiple locations in each of the rooms. Correlation analysis and statistical modeling identified significant relationships among intelligibility scores of the listeners with the physical measurements made in the room. Special attention was given to the architectural characteristics of the rooms in the analysis. The variations in intelligibility scores and the physical measurements among the different rooms and within each of the rooms are presented. [Work supported by NSF.]

10:45

4AA5. Modern acoustic measurements on three types of concert platforms. John P. M. O'Keefe and Marc Bracken (Barman Swallow Assoc., 1 Greensboro Dr., Rexdale, Ontario M9W 1C8, Canada)

Recent studies, particularly those by Gade and Naylor, have begun to quantify the acoustical requirements of performers on stage as expressed in measurable acoustic parameters. These parameters, originally based on laboratory findings, are now being measured in real concert halls. The purpose of this investigation is to add to this database by measuring the conditions on three different types of concert platforms: A proscenium theatre stage, a multipurpose room with a reflector above the stage, and finally, a traditional concert hall stage. Measurements in the multipurpose room will also quantify the effect of reflector height above the stage in terms of support (ST_{100}) and modulation transfer functions. Measurements in the proscenium theatre will quantify conditions on stage, in the orchestra pit, and between the stage and the pit.

11:00

4AA6. Experimental measurements of auditoria acoustic quality. A. Cocchi, P. Fausti, and M. Garai (Inst. di Fisica Tecnica, Univ. di Bologna, Viale Risorgimento 2, 40136 Bologna, Italy)

This paper illustrates an original implementation of Ando's theory for the evaluation of the acoustical characteristics of concert halls. In most cases, preliminary research is carried out on physical scale models in order to identify the temporal distribution of the various reflections into the room, or using a *ray tracing* program which can also compute other important acoustic descriptors. The following measurement procedure inside the hall is completely automatized: The data are acquired via an artificial head carrying two microphones in the place of the ears; the head is linked to a two-channel FFT analyzer that performs a digital analysis of the signals under control of a microcomputer that also completes some parts of the computation. This fast, *in situ* elaboration of the data, make it possible to carry out in a short time many measurements in different points of the hall and to obtain for each of them the four parameters of the Ando's theory (IACC, ITDG, T_{sub} , L_s). From these values, a *preference index* can be determined for each point inside the room. The above mentioned techniques have already been used to qualify and to compare different kinds of theatres and auditoria.

11:15

4AA7. Scale-model acoustical analysis of the Performing Arts Center at Swarthmore College. Ming Hin Soon and E. Carr Everbach (Dept. of Eng., Swarthmore College, Swarthmore, PA 19081-1397)

The Lang Performing Arts Center is a newly constructed facility that will house the Departments of Drama and Dance at Swarthmore College. The facility contains a 1000-seat theater with adjustable staging and a moveable concrete wall that can bisect the space into two separate theaters for simultaneous use. A 1/48 scale model of the theater has been constructed and an analysis of the model and a comparison of the model and measurements made in the theater itself are presented. Of special interest are some of the devices that were developed to provide acoustic excitation in the model and in the full-scale theater.

11:30

4AA8. Experimental evaluation of the backscattered polar response of sound absorbing, reflecting, and diffusing materials. Peter D'Antonio (RPG Diffusor Syst., Inc., 12003 Wimpleton St., Largo, MD 20772)

The sound that we hear in a room is a combination of the direct sound and indirect reflected sounds from room boundaries. Because the amplitude, directionality, and temporal distribution of the indirect reflections affect how one perceives the actual sound source, control of room reflections using absorption, reflection, and diffusion is a central consideration in acoustical design. Effective application of these ingredients is based on an understanding of the backscattered directional response for a given angle of incidence, angle of observation and frequency, i.e., the backscattered polar response. To evaluate the experimental polar response a testing technique using time delay spectrometry was developed [P. D'Antonio and J. Konner, AES Preprint 2295, 79th AES, New York (October 1985)]. The method allows the evaluation of the backscattered frequency response of a sound modifying material mounted on the boundary surface at any scattering angle, for a given angle of incidence. For sound absorbing materials, for example, these "directional" scattering coefficients augment the traditional random-incidence ASTM reverberation chamber data. The technique will be described and polar response data for sound absorbing, reflecting, and QRD⁹ diffusing panels will be presented. An alternative method using sound intensity will also be discussed and preliminary data presented.

11:45

4AA9. Acoustical design of the National Center for the Performing Arts of Peru. Carlos R. Jimenez-Dianderas and Manuel Panduro-Manrique (Garcilaso de la Vega 163, Salamanca de Monterrico, Lima 3, Peru)

The project is based on the design of a cultural complex for the National Center for the Performing Arts which includes a 1200-seat opera house, a 750-seat concert hall, a small auditorium for master classes, and an amphitheatre for 1500. The interest in this project came from the lack of cultural facilities in Peru. Furthermore, it is an established methodological alternative to the architectural acoustic design; so the Center is the result of the integrity of a two-design process combined into one architectural and acoustical design. The site for the project was determined from the evaluation of several possibilities through analysis of successive approximations. The analysis includes the urban acoustical study of the metropolitan area of Lima. The final decision about the site was done through acoustical measurements of urban noise. The acoustical design of the auditoria includes ray-diagram analysis, reverberation time, early decay time calculation, and final sound level in some positions.

Session 4EA

Engineering Acoustics and Structural Acoustics and Vibration: Wave Propagation in Elastic Media

Sung H. Ko, Chair

Naval Underwater Systems Center, Code 2133, New London, Connecticut 06320-5594

Chair's Introduction—8:30

Invited Papers

8:35

4EA1. Some experimental studies in elastic wave propagation. H. Kolsky (Div. of Appl. Math., Brown Univ., Providence, RI 02912)

A number of experimental studies of elastic wave propagation in solids, with which the author has been associated, are reviewed. These concern the propagation and reflection of mechanical pulses in elastic solids. In most of these studies, unexpected mechanical response has been observed. The problems discussed include that of a glass beam subjected to flexural loading [H. Kolsky, *Dynamic Crack Propagation*, edited by G. C. Sih (Noordhoff, Groningen, 1973), pp. 399–414; V. Kinra and H. Kolsky, *Eng. Fracture Mech.* **9**, 423–432 (1977); H. J. Schindler and H. Kolsky, *J. Mech. Phys. Sol.* **31**, 427–437 (1983)] and it has been shown that the time for the fracture to be completed is equal to the time it takes the extensional compressive wave generated by the growing crack to travel to the free ends of the beam and be reflected back as a tensile pulse. Another problem discussed is that of the elastic impact of a sphere on a plate. When the plate is sufficiently thick, i.e., its thickness is such that the sphere separates from the plate before reflected waves reach the region of contact. The problem, which was first treated by Hertz, has more recently been treated by S. C. Hunter [*J. Mech. Phys. Solids* **5**, 162 (1957)], who showed that about 1% of the energy is converted into elastic vibrations of the block. C. Zener [*Phys. Rev.* **59**, 669 (1941)] has shown that the low coefficients of restitution observed for thin plates can be explained in terms of flexural plate waves. More recently, M. G. Koller and H. Kolsky [*Int. J. Solids Structures* **23**, 1387–1400 (1987)] have studied the problem in detail in terms of the propagation and reflection of the waves generated by the impact and shown two unexpected phenomena, namely the propagation of a longitudinal disturbance at the *S*-wave velocity, and an enhancement of amplitude on reflection of an outgoing *P* wave. Other studies which will be discussed are the change in pulse shape which occurs when a unidirectional pulse is reflected at the boundary between two noncollinear rods [J. P. Lee and H. Kolsky, *J. Appl. Mech.* **39**, 809–813 (1972)], and between a rod and a half-space [S. Boucher and H. Kolsky, *J. Acoust. Soc. Am.* **52**, 884–898 (1972)].

9:00

4EA2. Analytical and experimental results for scattering of ultrasound by surface-breaking cracks. J. D. Achenbach (Ctr. for Quality Eng. and Failure Prevention, Northwestern Univ., Evanston, IL 60208)

Mathematical modeling of ultrasonic wave scattering provides valuable quantitative information for methods to detect and characterize cracks. Even though both the geometrical configuration and the process of ultrasonic wave propagation must be simplified, the characteristic features of the scattering phenomenon can be maintained. Results obtained by the modeling approach aid in the design of efficient testing configurations, as well as in the interpretation of experimental results and field data. Once a mathematical model has been verified by comparisons with a sufficiently wide range of experimental data, representative synthetic data can be generated with little effort. This is useful for the generation of a knowledge base for an expert system and for the development of data to train a neural network. The analytical results presented here for surface breaking cracks have been obtained by the use of the boundary integral equation/boundary element method. Since the elastodynamic Green's function is for an elastic half-space is complicated, it was found advantageous to use the full space Green's function. The use of that function implies that the system of boundary integral equations includes equations over the free surface of the half-space. The calculated results show excellent agreement with experimental results that will also be presented. [Work supported by ONR.]

4EA3. Underground sound: Applied seismic waves. J. E. White (Geophys. Dept., Colorado School of Mines, Golden, CO 80401)

The term underground sound is meant to encompass the instrumentation, observational methods, and interpretive techniques that have been developed for the utilization of seismic waves in the earth, quite analogous to the field of underwater sound. Wave propagation in the shallow crust of the earth is more complex than waves in water, simply because of the presence of shear waves in addition to compressional waves. Either type of wave is severely attenuated in traveling through earth materials, compared with similar distances of travel through the ocean. Finally, many rocks are anisotropic in average properties, resulting in further complexities. In exploration for petroleum, remarkably detailed images of geologic boundaries have been achieved by processing observed data so as to enhance reflected compressional waves and suppress shear waves and reverberations. Alternatively, shear-wave reflections can be observed by using a vibratory source which radiates strong shear waves and processing the data so as to suppress compressional "noise" waves. Used together, the two reflection methods yield lithologic information, such as distinguishing between gas saturation and water saturation in a porous rock. For soils and near-surface rocks, shear-wave speeds are directly useful for evaluation of building foundations and dam sites. Attenuation of seismic waves is both a fortunate characteristic and a strict handicap. If attenuation were quite small, multiple reflections among layers would result in uninterpretable reverberations. Since the actual attenuation increases with frequency, the resulting loss of high frequencies places a severe limit on the resolution obtainable with seismic waves. The presence of attenuation requires that the wave speed depend on frequency. When averaged over distances of a few meters, many geologic formations behave as "homogeneous" layers. However, the average properties may depend on direction and the layers must be recognized as anisotropic solids. Such anisotropy may be due to fractures caused by previous episodes of deformation, due to unequal stresses existing at present, or due to fine layering. Anisotropy clearly complicates seismic data processing. Because so many measurements are made in oil wells and test boreholes, suitable sources and detectors of seismic waves have been developed and the coupling of seismic waves to a fluid-filled borehole has been treated mathematically.

9:50

4EA4. Wave propagation and scattering in chiral elastic media. Vijay K. Varadan and Vasundara V. Varadan (Res. Ctr. for the Eng. of Electron. and Acoust. Mater., Penn State Univ., 149 Hammond Bldg., University Park, PA 16802)

Optical activity is exhibited by media whose molecular configurations are handed or chiral. Since the geometry is the basis for chirality, it can be probed by transverse waves, but not by longitudinal waves. Composite elastic media can also be made chiral because the elastic field in solids consist, in general, of both longitudinal and transverse components. The displacement field u in such isotropic, noncentrosymmetric (chiral) elastic solid has to be supplemented by the independent microrotation field ϕ . Six different wave numbers are possible in chiral solids. Of these, two represent longitudinal fields and the remaining four are circularly polarized. Composites can thus be tailor-made by suitably designing the microstructure and volume fraction of chiral elements. In this talk, the theoretical foundation for wave propagation and scattering in chiral elastic solids will be outlined. Pertinent experimental results on scattering of acoustic waves by chiral elastic composites containing piezo-chiral elements will also be presented.

10:15

4EA5. Transient and steady-state scattering of acoustic waves from elastic objects in fluids, and of elastic waves from inclusions in solid media—Direct and inverse scattering aspects. G. C. Gaunaurd (Naval Surface Warfare Ctr., Res. Dept. (R-42), White Oak, Silver Spring, MD 20903-5000)

This paper considers scattering problems associated with: (a) the simpler case of acoustic wave scattering by an elastic body in a fluid, and (b) the more complex situation of elastic waves scattered by a spherical (fluid or elastic) inhomogeneity in a solid host medium. The analysis is first presented in the frequency domain, in general, for all frequencies. Later, particular emphasis is given to the midfrequency range: $5 \lesssim ka \lesssim 25$, which coincides with the "resonance region" of impenetrable scatterers. It is in this region that a number of simplifying approximations valid at low or high frequencies do not hold and the analysis is hardest. The above two cases were the ones originally treated by the RST. Since there are also resonances in the high-frequency region and in the Rayleigh region ($ka \lesssim 5$), the concept of "resonance region" has to be broadened for penetrable scatterers. However, resonances in the Rayleigh region have relatively simple (viz., either multipole or curvature) origins that will be reviewed here. The analysis is then extended (still *direct* scattering) to the time domain. The target's transient responses to selected types of incident pulses are computed, measured, and interpreted. Finally, selected aspects of the *inverse* scattering problem are considered. Also examined (in various cases) is how certain features in the target's time or frequency "signature" are related to specific physical target characteristics (viz., shape, composition,...) that permit its remote and unambiguous classification. [Work supported by ONR and NSWC.]

10:40

4EA6. Elastic waves in rods and optical fibers. R. N. Thurston (Bellcore, Red Bank, NJ 07701-7040)

This talk reviews elastic waves in rods, with mention of their possible importance in optical fibers. The formal solution of the boundary value problem of the three-dimensional theory of elasticity for waves in rods was given over a century ago by Pochhammer and Chree. It was only about 50 years ago that important computations were done that enabled calculated results to be compared with experimental data for longitudinal waves [D. Bancroft, *Phys. Rev.* **59**, 588-593 (1941)] and flexural waves [G. E. Hudson, *Phys. Rev.* **63**, 46-51 (1943)]. The modulation or scattering of light by acoustic waves in optical fibers can be either harmful as in catastrophic stimulated backward Brillouin scattering [E. P. Ippen and R. H. Stolen, *Appl. Phys. Lett.* **21**, 539-541 (1972)] or it can be put to practical use as has been attempted in devices such as acoustically mode-locked fiber lasers [Phillips *et al.*, *Opt. Lett.* **14**, 680-682 (1989)].

11:05

4EA7. Application of elastomeric material to the reduction of turbulent boundary layer pressure fluctuations. Sung H. Ko (Naval Underwater Systems Ctr., New London Lab., New London, CT 06320)

Turbulent boundary layer pressure fluctuations can be reduced by either spatial filtering through the use of a finite hydrophone or a hydrophone array or by filtering through direct path attenuation. In general practice, various configurations of hydrophone arrays are embedded within a layer of elastomer, thus reducing the turbulent boundary layer pressure fluctuation. The direct path attenuation depends on the elastomer layer thickness, the shear wave speed, and the loss factor associated with the shear wave in the elastomer layer. In the filtering process, the mechanism that controls the direct path attenuation is most important. The theoretical model considered in this paper is a plane elastomer layer backed by an infinite plate with finite thickness; the other side of the layer is exposed to turbulent flow. A theoretical analysis is presented for the development of the transfer function that determines the amount of direct path attenuation. The results presented are numerically calculated transfer functions and noise reductions for various parameters related to direct path attenuation.

WEDNESDAY MORNING, 1 MAY 1991

PRATT A & B, 9:00 TO 11:40 A.M.

Session 4MU

Musical Acoustics: Bowed Strings: Honoring Carleen Hutchins, Part 1

Gabriel Weinreich, Chair

Randall Laboratory of Physics, University of Michigan, Ann Arbor, Michigan 48109

Invited Papers

9:00

4MU1. Carleen: An appreciation. Gabriel Weinreich (Randall Laboratory of Physics, University of Michigan, Ann Arbor, MI 48109-1120)

Carleen Hutchins' own accomplishments, as well as the work she has stimulated others to do, not only throws much light on the nature of the art/science interface but also helps to illuminate the riches that can be mined on both sides of it.

9:10

4MU2. New aspects of the violin vibrato. Jürgen Meyer (Physikalisch-Technische Bundesanstalt, Bundesallee 100, D-3300 Braunschweig, Germany)

Usually the vibrato played on a violin is considered as a frequency modulation caused by the rolling motion of the touching finger; typical values are about 4.5-8 Hz for the vibrato frequency and up to $\pm 40\%$ and even more for the width of the vibrato. But the high number of more or less sharp resonances of the violin leads to an amplitude modulation of the partials too. Because of the unharmonic positions of these

resonances some overtones are modulated in phase and other ones in antiphase. At higher frequencies individual partials fluctuate over the peak of a resonance or through the gap between two resonances and show an amplitude modulation of twice the vibrato frequency. When lower partials fluctuate in the order of 6 dB, components about 5000 Hz may reach an amplitude modulation of 20 dB or in extreme cases up to 35 dB. Because of the directivity these values depend on the direction of sound radiation, too. It may be a criterion for the sound quality of a violin, whether there are such enormous fluctuations or not. In larger rooms reflections having different delay times average the amplitude modulation and lead to filled frequency bands for the partials.

9:40

4MU3. The linear and nonlinear vibrations of a stretched string and the relevance to stringed musical instruments. C. E. Gough (School of Physics and Space Res., Univ. of Birmingham B15 2TT, UK)

For small amplitudes of string vibration, the modes of a stretched string are shown to be equivalent to the normal modes of a coupled oscillator problem, the coupled oscillators being the string itself and the vibrational modes of the body on which the string is supported. The coupling at the bridge splits the degenerate modes of the ideal string, with a degree of perturbation that depends on the strength of the coupling and the damping of the various modes involved. Under extreme conditions this can lead to a splitting of the modes resulting in the famous wolf-note on bowed instruments, which was first studied by Raman and later by Schelling. At large amplitudes, the nonlinear coupling of the orthogonal transverse modes will be shown to result in a precession in the plane of polarization of the now elliptically polarized transverse modes, which could lead to variations in amplitude of emitted radiation of any stringed instrument. In this talk, the theoretical predictions for both linear and nonlinear string vibrations will be briefly outlined and measurements will be described for idealized experimental situations and for instruments of the violin family.

10:10

4MU4. Observations during a design and test program for bowed instrument strings. Norman C. Pickering (23 Culver Hill, Southampton, NY 11968)

During the past 3 years, the author has participated in a program to review and redesign the entire bowed string line of a major manufacturer, using new materials and manufacturing techniques in addition to traditional designs and methods. Computer programs have been created, both for design and for control of quality. In testing hundreds of strings, some anomalies have been seen, including persistent whirl modes and variation of frequency and damping with amplitude. Attempts have been made to ascertain whether such conditions may affect musical performance. The action of rosin has been studied, and temperature at the bow-string interface measured with a calibrated infrared video camera. The greatest difficulty has been correlating opinions of musicians with actual measurements, a problem that is not unexpected in musical acoustics.

10:40

4MU5. On violin plate and corpus tuning. Oliver E. Rodgers (Mech. Eng. Dept., Univ. of Delaware, Newark, DE 19711)

The tuning of violin free plates in the manner advocated by Carleen Hutchins is now established practice. One can be sure of producing an instrument with good sound. However, the way to an instrument that has excellent sound is still elusive and seems to depend on details of construction that continue to elude investigators. This paper describes some of the existing clues that identify important details of violin design and construction that may lead to further progress toward achieving superior performance, including adjustment of the assembled corpus and complete instrument, guided by acoustical measures. Some suggestions are made of directions for further research on violins and the larger stringed instruments. Only measurement of the sound output of test instruments, tempered by what the human ear hears, will provide the information necessary to make additional progress.

11:10

4MU6. A preliminary examination of the out-of-plane motion of the violin bridge. Kenneth D. Marshall (Uniroyal Goodrich Tire Co., 9921 Brecksville Rd., Brecksville, OH 44141)

The vibrations of violin strings excited by bowing or plucking are mainly in the transverse and vertical directions, and are communicated to corpus through the rocking and vertical motions of the bridge. These factors have been extensively studied, but much less is known about how the longitudinal and torsional vibrations of the string, and the out-of-plane motions of the bridge, affect the sound radiated by the instrument. This talk will briefly review what has been learned to date about the out-of-plane vibrations of the bridge. Information is then presented showing: (1) the in-plane (y and z direction) motions of the bridge generally related at low frequency but quite different at higher frequency, and (2) the out-of-plane (x

direction) bridge vibrations, distinctly different from the in-plane, and in several instances providing a negative resistance that may absorb energy from the vibrating strings. It is concluded that three-dimensional bridge motion should be considered to provide an improved understanding of violin dynamics.

WEDNESDAY MORNING, 1 MAY 1991

INTERNATIONAL E, 8:00 TO 11:05 A.M.

Session 4NS

Noise and Psychological and Physiological Acoustics: Hearing Loss Prevention and Compensation

Edwin H. Toothman, Cochair

Bethlehem Steel Corporation, 701 East Third Street, Room 169 SGO, Bethlehem, Pennsylvania 18016-7699

Larry D. Royster, Cochair

*Department of Mechanical and Aerospace Engineering, North Carolina State University,
Raleigh, North Carolina 27607-7910*

Chair's Introduction—8:00

Invited Papers

8:05

4NS1. Hearing loss compensation from the worker viewpoint—Technical aspects. Robert D. Bruce
(Collaboration in Science and Technology Inc., 15835 Park Ten Pl., Ste. 105, Houston, TX 77084-5131)

Workers view hearing loss and compensation for hearing loss from a very personal perspective. Although agencies administering compensation programs may have made significant efforts to take into consideration the total impact of hearing loss on an individual when standards were set for compensation, it is clear that there is a discrepancy between the amount allowed under existing compensation systems and the value that an individual places on his/her hearing. In an effort to understand the worker viewpoint, this paper will compare the loss of hearing (and its acoustic and nonacoustic impacts) with other accidental injuries in industry, review some of the existing compensation systems (state, federal, and private) for hearing loss and for other injuries, review the impairment concept as it relates to hearing loss, summarize the adjustments in compensation that have occurred during the past 20 years, and present a forecast for the future.

8:30

4NS2. Hearing loss compensation from the worker viewpoint—Legal aspects. M. Juliet Lawson (Attorney,
Richard F. Scruggs, P.A., 734 Delmas Ave., Pascagoula, MS 39567)

Workers in the United States who are employed by individuals not engaged in maritime work can recover for occupational hearing loss under state worker's compensation statutes. If the employer is a maritime employer, the worker can recover under the federal Longshore and Harbor Worker's Act for occupational hearing loss. The worker may also have several causes of action against the manufacturer of the product or products that he used or worked around. The worker may allege that the manufacturer was negligent and breached a duty to the claimant to exercise the highest standard of care in the manufacture of its products and/or the worker may claim that the manufacturer is strictly liable for any damage that the worker suffered because the product was distributed in a defective condition unreasonably dangerous to intended and foreseeable users or bystanders. There may also be a claim for breach of an implied or express warranty of fitness and merchantability.

4NS3. Hearing loss compensation—The defense prospective. Richard W. Scheiner (Semmes, Bowen & Semmes, 250 W. Pratt St., Baltimore, MD 21201)

The lecture will focus on the current state of hearing loss claims in the workplace. The discussion will include the prevalence of hearing loss claims in heavy industry, the factors influencing the volume of claims including the influence of unions and organized labor, the feeling that these claims yield easy money, and the effect of the depressed economy. Information will be given regarding the cost factor of these claims to the employer and industry. The lecture will include an overview of the Longshore & Harbor Workers' Compensation Act (LHWCA) and certain portions of that law which play a strong part in the resolution of hearing loss claims including the last injurious exposure rule, the no apportionment rule, the effect of advancing age and/or retired workers on hearing loss claims, and the virtual elimination of the defenses of the statute of limitations and notice. Parallel comparison will be made to state workers' compensation acts focusing primarily on the State of Maryland. A review of the many factors which favor the claimant's position in hearing loss cases will be discussed. In addition, the lecture will review various legal tactics and theories used in defending hearing loss claims including some recent decisions under the Longshore & Harbor Workers Compensation Act which affect and may ultimately change defense tactics.

9:20

4NS4. Hearing loss compensation from the corporate viewpoint. M. Russell Guy (Law Dept., Bethlehem Steel Corp., Rm. 1968, Martin Tower, Bethlehem, PA 18016-7699)

Manufacturing operations are located mainly in three states which present widely varying requirements for hearing loss compensation; the compensation case experiences in these three jurisdictions will be discussed. The relationship of hearing loss compensation costs to total compensation costs will be presented. An explanation will be given on the approach utilized to address/handle hearing loss claims. The importance of preventing occupational hearing loss will be emphasized.

9:45

4NS5. Preventing compensable on-the-job noise-induced hearing loss. Larry H. Royster (Dept. of Mech. and Aerosp. Eng., NCSU, Raleigh, NC 27695-7910) and Julia Doswell Royster (Environmental Noise Consultants, Inc., P.O. Box 30698, Raleigh, NC 27622-0698)

To prevent a significant on-the-job noise-induced hearing loss, management must ensure that hearing conservation program (HCP) activities include the following efforts (as appropriate): pre- and post-employment audiograms, annual audiograms for employees with TWAs of 85–99 dBA and semiannual audiograms for employees with TWAs of 100 dBA and higher, use of a threshold shift criterion more strict than the OSHA STS to flag the most noise-susceptible employees, documentation of adequate hearing protector fitting, issuing, replacement, and user training, and annual updating of auditory history information. The HCP's level of effectiveness should be evaluated annually using the recommendations of ANSI S12/WG12; if the program is found ineffective, corrective measures should be implemented immediately. When employees file for compensation, the appropriateness of their claims is evaluated by taking into consideration the level of effectiveness and completeness of the HCP, comparing their TWA exposure level against the real-world protection capabilities of the hearing protection utilized, and comparing their hearing losses against the range of hearing loss predicted using the ISO 1999 (1990) model for a noise-susceptible population with the same exposure. If an employee's hearing loss is judged to be non-work-related, then all company personnel and other professionals who will be involved in the judicial process should be adequately trained in preparation for the hearing or trial.

10:10

4NS6. Hearing loss prevention—Worker's responsibilities. James E. Detwiler (CIH Environmental, Inc., 12 Victory Circle, Reading, PA 19605)

One element of hearing loss prevention that management often feels it cannot effectively control is worker's actions and responsibilities. The solution to this problem is the development and enforcement of effective management-backed employee policies covering each hearing conservation issue. It is absolutely vital that all such policies be in writing and that they be clear, reasonable, sensible, and enforceable. Principal program elements that need written policies are monitoring, engineering and administrative controls, personal hearing protection devices, training and education, audiometric testing, and supervisors' responsibilities in program administration. Under each program element, one or more specific policy objectives should be spelled out, with each objective followed by a detailed policy statement of the actions or behavior required to meet that objective. Numerous examples of policy objectives and statements were written for each of the principal program elements. Finally, for the policies to be enforceable, there must be in place a fair and consistent system of discipline. Properly written and enforced policies are not designed to restrict the individual, but to protect him by assuring him safe working conditions and equal treatment with his fellows.

10:35

4NS7. Mining occupations with high incidence rates for noise-induced hearing loss. Michael P. Valoski (Mine Safety and Health Admin., Pittsburgh Health Technol. Ctr., 4800 Forbes Ave., Pittsburgh, PA 15213)

The Mine Safety and Health Administration (MSHA) requires that mine operators report to MSHA all cases of diagnosed or compensated occupational illness. Listed among the examples of reportable occupational illnesses is noise-induced hearing loss (NIHL). All cases of occupational hearing loss reported to MSHA between 1986 and 1989 (1288 cases) were examined. After separating the hearing loss cases by causal factor, a population of 1264 cases of NIHL remained for retrospective study. This study was conducted to identify those occupations which had substantially larger numbers of NIHL cases or higher incidence rates of NIHL. The NIHL cases were separated by coal (1030 cases) and metal/nonmetal mining (234 cases). The population data were obtained from a Bureau of Mines database. Since this database only contained data on occupational groups, not individual occupations, the incidence rates were calculated for occupational groups. The continuous miner and related machine operator occupational group had both the most cases of NIHL and the highest incidence rate for coal mining. While for metal/nonmetal mining, the mechanic-welder-oiler-machinist occupational group had the most cases of NIHL and the shuttle car-tram operator occupational group had the highest incidence rate. The work location which had the highest incidence rate of NIHL

was preparation facilities for coal mining and underground mines for metal/nonmetal mining.

10:50

4NS8. Maximizing communication ability in the selection of hearing protective devices. Dennis Williams (Dept. of Commun. Disord., Penn State Univ., 3 Moore Bldg., University Park, PA 16802) and Kevin Michael (Michael Associates, Inc., 246 Woodland Dr., State College, PA 16802)

The most common procedure for selecting hearing protective devices (HPDs) is the Noise Reduction Rating (NRR). The NRR can be misleading and often results in the selection of HPDs that either under- or over-protect the wearer. The wearer of an under-protective HPD is likely to experience noise-induced hearing loss. The use of over-protective HPDs can needlessly reduce speech communication, the ability to hear warning signals, and the ability to recognize important machine sounds. Workers fit with over-protective HPDs are also less likely to properly wear their HPDs. An articulation index (AI) based method of ranking HPDs for use in specific noise exposures was developed in conjunction with the American Iron and Steel Institute (AISI). The calculations used in the procedure incorporate the attenuation characteristics of the HPD, the spectrum of the noise source, the hearing thresholds of the wearer, the nonlinear growth of masking, and the predicted level of speech in noise. The HPD rankings are intended to insure that adequate attenuation and optimum communication ability is achieved.

WEDNESDAY MORNING, 1 MAY 1991

LIBERTY B, 7:55 A.M. TO 12:00 NOON

Session 40C

Acoustical Oceanography: Open Workshop on Remote Sensing of Sediment Properties by Measurements on or Near the Seafloor

George V. Frisk, Chair

Bigelow 208, Woods Hole Oceanographic Institution, Woods Hole, Massachusetts 02543

Chair's Introduction—7:55

Invited Papers

8:00

40C1. Seafloor shear wave velocity structure determined from interface wave dispersion. LeRoy M. Dorman, Anthony E. Schreiner (Marine Physical Lab., Scripps Inst. of Oceanography, UCSD, La Jolla, CA 92093-0215), and L. D. Bibee (Code 360, Naval Oceanographic and Atmospheric Res. Lab., NSTL Station, MS 39529)

The propagation of interface waves in the seafloor waveguide is primarily controlled by the sediment shear velocity, which increases rapidly with subbottom depth. This gradient causes the propagation velocity to be strongly frequency dependent and this dispersion allows one to infer the shear velocity structure from propagation velocity measurements. These waves were excited with small explosions on the seafloor and the dispersion was measured over distances of a few hundred meters, observing seafloor motion on ocean bottom seismographs (OBSs). It is common to observe these waves in the 0.5–5 Hz frequency range, even from explosions with bubble frequencies in the 100- to 400-Hz range. Typically several modes are seen and shear velocity in the 1- to 30-m depth range can be recovered. The attenuation is high ($Q = 25$) in the very near

surface, consequently, modes with large energy just below the surface suffer severe attenuation, while modes with little energy in the highly absorbing regions (higher modes and all modes whose phase velocity is near that of water) show Q up to 500. [Work supported by ONR.]

8:20

40C2. Sediment properties derived from shear and interface waves. John I. Ewing, George H. Sutton (W.H.O.I., Woods Hole, MA 02543), and Jerry A. Carter (S.A.I.C., Center for Seismic Studies, Arlington, VA 22209)

Bottom-mounted sources and receivers have been used to measure shear wave velocity and attenuation in sediments of the U.S. East Coast continent shelf. Experimental results are compared with borehole and other subseafloor geologic information. Shear sources were designed primarily for generating SH waves but also produced SV and P waves. Receiver nodes containing orthogonal geophone or accelerometer sensors, plus hydrophone, provided four-component data, each component producing unique information on wave type, velocity/attenuation structure, scattering, lateral heterogeneity, and anisotropy. Measurements were made with two systems; one with a large source and 4–8 m sampling interval to a maximum range of 200 m, the other with a smaller source and 1-m sampling to a range of 30 m. Velocity and attenuation are estimated by matching recorded data with full-waveform synthetic seismograms. [Work supported by ONR.]

8:40

40C3. Remote sensing of sediment properties using seafloor arrays. Robert D. Stoll (Lamont-Doherty Geological Observatory of Columbia Univ., Palisades, NY 10964)

The placement of both source and receiver on the seafloor yields significant advantages when measuring the geoacoustic properties of near-bottom sediments. Moreover, by using multiple receivers in an array and specifically designed sources that focus energy into the bottom and minimize the water-borne "noise," it is possible to measure both p - and s -wave properties in significant detail. In this paper a detailed example of this technique illustrating equipment design, data acquisition, and inversion of data to obtain a geoacoustic model is presented. Results show that the horizontal component of motion contains important information that cannot be derived easily from hydrophone or vertical motion data. In addition, both near-field and far-field data are shown to be complementary in determining near-bottom, high-resolution models.

Contributed Paper

9:00

40C4. Application of remote sensing and *in situ* measurements of ocean sediment properties to the prediction of acoustic propagation loss. T. Akal, A. Caiti, F. Ingenito, and A. Kristensen (SACLANT Undersea Res. Ctr., APO, NY 09019)

An experiment was conducted in the Adriatic Sea to evaluate the ability of geophysical models of the ocean bottom, constructed by the application of *in situ* and remote sensing measurement techniques, to predict acoustic propagation loss. Short-range wide-angle measurements

of acoustic reflection from the bottom and measurements of the dispersive characteristics of seismic interface waves at the water-sediment boundary are used to estimate the P - and S -velocity profiles in the upper 30 to 50 m of the bottom sediment. Pointwise *in situ* and core measurements were taken at the same site. Simultaneously, broadband propagation loss measurements were made. Geophysical models were constructed from the bottom measurements and used as inputs to state-of-the-art acoustic models to predict propagation loss. Satisfactory agreement between measured and predicted propagation loss was obtained, indicating the general validity of the procedure.

9:15–9:30

Break

9:30

4OC5. Experimental verifications of bottom shear modulus profiler (BSMP) method. Tokuo Yamamoto (Geo-Acoustics Lab., University of Miami RSMAS, 4600 Rickenbacker Cswy., Miami, FL 33149) and Mark Trevorrow (Institute of Ocean Sciences)

By measuring the response of a seabed to the pressure forcing of traveling ocean waves, the seabed shear modulus profile can be extracted by solving a geophysical inverse problem [Yamamoto and Torii, *Geophys. J. R. Astron. Soc.* **85**, 413–431 (1986)]. The experimental results as favorably compared with geological boreholes have been reported [Yamamoto *et al.*, *Geophys. J. Int.* **98**, 173–182 (1989)]. ONR multi-institutional experiments were recently conducted to compare the various experimental methods for measuring the seabed shear properties at several sites on the continental shelf of the northeastern United States. Comparisons between the BSMP results and the *SH* wave refraction results by J. I. Ewing will be reported in this paper. Generally, good agreements between the two methods are obtained. BSMP method penetrates 200 m below seafloor (b.s.f.) at resolution of a few meters while the *SH* wave seismic method penetrates 50 m b.s.f. at approximately the same resolution. As an example application of the BSMP results to modeling, comparisons between model predictions (using BSMP data), and propagation experiments of seismic waves and acoustic waves in the ocean by W. Carey and the present authors are also made, resulting in favorable agreements. [Work sponsored by ONR.]

9:50

4OC6. A pseudo-underway geophysical technique for quantifying seabed sediment properties. Angela M. Davis (School of Ocean Sciences, Univ. Coll. N. Wales, Menai Bridge, Wales LL59 5EY, UK)

A towed geophysical device has been developed that will allow the rapid quantification of seafloor sediment properties for engineering and other purposes. The seabed hardware consists of seismic sources, a focused electrode pad (all sledge mounted), and a string of gimballed triaxial geophones. The seismic sources are impulsive devices that are essentially electromagnetic hammers that can preferentially generate shear or compressional waves dependent on their mode of operation. Travel time relationships obtained for seismic waves propagating through the sediment body to the geophone array allow the velocity structure to be defined to a depth dictated by the maximum receiver offset. Of particular interest to the engineers are the velocity gradient effects that for shear waves are indicative of the seabed sediment rigidity. The device can be used to construct distribution maps of geophysically related physical properties of the sediment body, and their variation with depth. Surveys carried out with the device in well-documented test-bed sites have produced significant correlations between the geophysical parameters and the known sediment variability.

10:10

4OC7. Near surface measurements of sediment geoacoustic properties using *in situ* probes. Michael D. Richardson (Naval Oceanographic and Atmospheric Res. Labs., Stennis Space Center, MS 39529-5004)

In situ and laboratory measurements of sediment geoacoustic properties together with sediment physical property measurements were made over the range of sediment types commonly encountered on continental shelves. *In situ* compressional wave velocities ranged from 1464 m/s in soft silty-clay sediments to as high as 1989 m/s in gravels. Velocities in sands were 1600–1700 m/s. Compressional wave attenuations (measured at 58 kHz of 4–12 dB/m) were common in soft silty-clay sediments with higher attenuations of near 30 dB/m measured at sandy sites. *In situ* shear wave velocities ranged from 16 m/s in soft silty-clay sediments to 90 m/s in hard packed fine sands. Differences among *in situ* and laboratory values of sediment geoacoustic properties are discussed. Empirical relationships between sediment geoacoustic and physical properties are presented.

Contributed Papers

10:30

4OC8. Bottom backscattering modeling and model/data comparison for 100–5000 Hz. Pierre D. Mourad, Peter H. Dahl, and Darrell R. Jackson (Appl. Phys. Lab., HN-10, Univ. of Washington, Seattle, WA 98195)

A low-frequency bottom backscatter model that includes both surface and (shallow) volume contributions is described. It is assumed that these different scattering mechanisms are uncorrelated. The surface

model [Jackson *et al.*, *J. Acoust. Soc. Am.* **79**, 1410–1422 (1986)] consists of Kirchhoff theory for large grazing angle contributions, spliced into composite roughness theory for backscatter from mid and low grazing angles. This part of the model is applied to isotropic, Gaussian rough surfaces whose surface height spectra are described by a two-parameter power law. These parameters are fixed by previous, high-frequency applications of this surface roughness theory. The volume scattering contribution is based upon an analytic theory for backscatter from uncorrelated, uniformly distributed point scatterers embedded within an upward refracting, constant density, “fluid” sediment layer.

For this part of the model, the water/sediment interface is smooth, with discontinuities in compressional sound speed and density. The model predictions are compared with backscatter observations that have supporting geoacoustical data. By using these geoacoustical measurements along with standard bounds in the literature, the model contains only one free parameter: a quantity proportional to the effective volume scattering cross section per unit volume of sediment. [Work supported by NAVOCEANO and APL.]

10:45

4OC9. High-frequency acoustic penetration of sandy ocean sediments. Nicholas P. Chotiros and Michael L. Ramaker (Applied

Res. Labs., The University of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

High-frequency acoustic signals, from 5 to 80 kHz, were projected from a point in the water to an array of hydrophones in a sandy sediment. The data were processed to yield acoustic wave speeds and directions. For normal and near-normal incidence, the results are in good agreement with liquid-viscoelastic solid propagation models. For grazing angles less than the critical value, the results cannot be explained in terms of a viscoelastic medium; a slow wave is observed that can only be explained in terms of Biot's theory. [Work supported by ONT under NOARL management.]

11:00–11:15

Break

11:15–12:00

Panel Discussion

PANEL MODERATOR: George V. Frisk

PANEL MEMBERS: LeRoy M. Dorman
John I. Ewing
Robert D. Stoll
Tokuo Yamamoto
Angela M. Davis
Michael D. Richardson

WEDNESDAY MORNING, 1 MAY 1991

CARROLL, 8:00 A.M. TO 12:00 NOON

Session 4PA

Physical Acoustics: Photoacoustics and Laser Ultrasonics

Yves H. Berthelot, Chair

Department of Mechanical Engineering, Georgia Institute of Technology, Atlanta, Georgia 30332

Chair's Introduction—8:00

Invited Papers

8:05

4PA1. Laser generation and detection of ultrasound and applications. Jean-Pierre Monchalin (Indust. Mater. Inst., Natl. Res. Council of Canada, 75 Blvd. Mortagne, Boucherville, Québec J4B 6Y4, Canada)

Ultrasound is generally generated and detected by using piezoelectric transducers bonded or fluid coupled to the specimen under investigation. Such a method requires a good bond or precise transducer orientation, is essentially limited to samples with planar surfaces, and can hardly be applied at elevated temperature. These limitations are circumvented by using lasers to generate and detect ultrasound. This

technique, called laser ultrasonics, has the advantage of being noncontact, permits probing off curved surfaces and complex shapes and allows the generation and detection of low frequencies as well as high frequencies. In this presentation, the various physical principles used for laser generation and optical detection will be outlined. Several interferometric detection systems that have been developed in this laboratory will be reviewed and their merits and limitations for laboratory investigation or industrial inspection will be discussed. Several applications now being pursued will be presented, including various thickness gaging applications, the generation and detection of plate waves and their use for evaluating plate anisotropy, the determination of ultrasonic attenuation, and its use for evaluating steel grain size.

8:30

4PA2. The generation of narrow-band and directed ultrasound using spatially and temporally modulated arrays. J. W. Wagner, J. B. Spicer, and J. B. Deaton, Jr. (The Johns Hopkins Univ. Ctr. for Nondestructive Evaluation, Maryland Hall 102, Baltimore, MD 21218)

Owing to their generally poor sensitivity relative to conventional contact ultrasonic methods, laser-based ultrasonic systems have proven to be effective for only a limited range of applications, and only then with very careful and specific designs. Recent investigations at The Johns Hopkins University have been directed at the use of spatial arrays and temporal laser beam modulation to generate ultrasonic signals which are both narrow band and may be directed over some angle within the test piece. The degree to which sensitivity may be enhanced by these methods, however, is a strong function of the temporal and spatial nature of the laser array source as well as the acoustic mode one wishes to excite. For example, variations in the dimension of each element in an array, array spacing, and array source rise time may all affect dramatically the degree to which one is able to generate narrow-band signals for high-sensitivity detection. Ultrasonic directivity issues are also somewhat more complicated than they are represented in much of the current literature. In fact, the degree to which the ultrasonic energy may be directed by laser array sources of any type is a strong function of the wave shape generated by each element in the array. Consequently, individual features such as pulse height or pulse repetition, which may be derived by superposition of the signals from elements of an array, may be directed over a range of angles in a solid material while total far-field energy directivity may remain unchanged.

8:55

4PA3. Photoacoustic monopole radiation from laser irradiated fluids. G. J. Diebold, M. I. Khan, T. Sun, and S. M. Park (Dept. of Chem., Brown Univ., Providence, RI 02912)

The temporal profile of photoacoustic waves generated by irradiation of fluid bodies with laser light is determined by the wave equation for pressure. Consider excitation where the heating is described as a product of a spatial heating function multiplied by a delta function in time. Solution of the wave equation in one dimension shows that the deposition of heat in space is mapped directly into the time profile of the photoacoustic wave. In three dimensions, spherically symmetric deposition of heat gives a photoacoustic wave proportional to the product of the retarded time with a symmetrized spatial heating function. The wave equation also gives solutions for photoacoustic waveforms generated by long light pulses. In one dimension the acoustic wave is proportional to exciting pulse, in three dimensions the emitted wave is proportional to the first derivative of the exciting radiation, whereas in two dimensions the acoustic wave is a complicated function of time. For long light pulses, the time profile of a photoacoustic wave depends only on the dimension of the wave and the time dependence of the exciting radiation. Expressions for acoustic multipole radiation from irradiated bodies are derived as well. Acoustic waves generated by irradiation of fluids with the output of a Nd:YAG laser are compared with the theoretical results.

9:20

4PA4. Photoacoustic frequency-domain depth profiling of continuously inhomogeneous solids. Theory and quantitative profilometry of octylecyano-biphenyl (8CB) liquid crystals. Andreas Mandelis (Photoacoustic and Photothermal Sciences Lab. and Ontario Laser and Lightwave Res. Ctr., Dept. of Mech. Eng., 5 King's College Rd., Univ. of Toronto, Toronto, Ontario M5S 1A4, Canada), Els Scoubs (Katholieke Univ. Leuven, Leuven B-3030, Belgium), Samuel B. Peralta (Univ. of Toronto, Toronto, Canada), and Jan Thoen (Katholieke Univ. Leuven, Leuven, Belgium)

An application is presented of the Hamilton-Jacobi formulation of thermal-wave physics [A. Mandelis, *J. Math. Phys.* 26, 2676 (1985)] to the problem of photoacoustic depth profiling in inhomogeneous solids with arbitrary, continuously varying thermal diffusivity profiles. Simple expressions for the modulation frequency dependence of the photoacoustic signal in the case of exponential thermal diffusivity profiles are obtained, and a working general method for solving the inverse problem and obtaining arbitrary diffusivity depth profiles is demonstrated through computer simulations. The method was found to possess excellent profile reconstruction fidelity. As a first experimental application of the theory, an observed change in the photoacoustic signal frequency response upon the application of a transverse magnetic field across octylecyano-biphenyl (8CB) samples in the nematic phase at 37 °C is reported. The theory has given quantitative profiles of thermal diffusivity decreasing extending to 20–30 μm below the liquid crystal surface. These decaying depth profiles are qualitatively consistent with earlier photoacoustic temperature scans of liquid

crystals and are a measure of the extent of bulk reorientational effects due to the magnetic field, as well as the extent of the influence of the surface as a domain reorientation inhibitor in the kG range.

9:45

4PA5. Poly(vinylidene fluoride) transducers for the detection of laser-induced transients. Hans Coufal (Almaden Res. Ctr., 33/802, IBM Res. Div., San Jose, CA 95120-6099)

Poly(vinylidene fluoride) (PVDF), a ferroelectric polymer, lends itself to the design and implementation of ultrafast pyro- and piezoelectric transducers for the detection of laser-induced thermal and acoustic transients. Crucial for the performance of these transducers is the depth profile of the polarization in the PVDF films. Polarization profiles in various PVDF foils and the influence of the profile on the observed signals are discussed. Applications of PVDF foils in transducers for thermometry and calorimetry of laser-induced thermal transients and in transducers for bulk and surface acoustic waves are then reviewed. Finally the potential and recent applications of other ferroelectric polymers for NDE are discussed.

10:10-10:35

Break

10:35

4PA6. Laser generation of ultrasound for process control using optical fiber arrays. Charles Umeagukwu, Jacek Jarzynski, Nick de Ridder, and Jr-Syu Yang (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332)

This paper reviews recent work on the use of optical fiber arrays to enhance laser generation of ultrasound. Experimental and numerical directivity patterns are presented, obtained using optical fiber arrays to generate longitudinal, shear, and surface waves. Comparisons of the directivity patterns for a single light source and for the fiber array will be presented and discussed for each of the above waves. Also shown will be some experimental results on array gains. This discussion will be limited to sound generation in the thermoelastic (linear) range. The optical fiber array can be used to control both the directivity and the type of elastic wave generated by the laser light. The noncontact fiber array generation can be combined with a noncontact EMAT or laser Doppler receiver to achieve a system for ultrasonic on-line inspection and control of manufacturing processes. The above technique for sound generation and reception is particularly useful in hostile and hard to reach environments. [Work supported by the National Science Foundation.]

Contributed Papers

11:00

4PA7. Deconvolution problems in a pulsed photoacoustic measurement of methane vibrational lifetimes at 77 K, 0.01 Torr. J. A. Burt, K. Yang (Physics Dept., York Univ., 4700 Keele St., Downsview, Ontario M3J 1P3, Canada), R. N. Halthore, J. E. Allen (NASA/Goddard), J. J. Caldwell, and P. Delaney (ISTS/SAL)

A conventional gas phase photoacoustic measurement has proved to be less than straightforward due to difficulties associated with deconvolution of the microphone (Bruel and Kjaer) response from the vibrational quenching lifetimes measured for methane in various binary buffer gases. The application of the lifetime data is to the aeronomy of Jupiter and preliminary results have been reported [Halthore *et al.*, "Implications of the methane relaxation time measurements for the atmospheres of outer planets and Titan," 21st Ann. Mtg. Div. Planetary Sci., AAS, 1989]. However, when extensive, accurate recordings were obtained this summer, a systematic error was noted that was attributed to convolution of the microphone's impulse response into the molecular relaxation dynamics. For this reason, it is felt that cw chopped laser excitation would be superior to the Nd-YAG pulsed laser excitation presently used to pump the asymmetric stretch mode of methane at 3.3 μm . [Work supported by the Institute for Space and Terrestrial Physics, Space Astrophysics Laboratory, Toronto.]

11:15

4PA8. Noncontacting surface wave measurements on spherical surfaces. Kenneth L. Telschow, Layman A. Lott, and Gregory W. Characklis (Idaho Natl. Eng. Lab., EG&G Idaho, Inc., Idaho Falls, ID 83415-2209)

Laser techniques have simplified the task of performing ultrasonic measurements on materials in nonplanar shapes, such as spheres and cylinders, since no physical contact is required. The propagation of surface waves on curved surfaces has been measured by using a pulsed laser source and a phase insensitive Fabry-Perot interferometer for detection. This detection method allows nonspecular surfaces and optically penetrating materials, such as ceramics, to be probed. Resonant and nonresonant conditions on nonmetallic spheres of varying size have been recorded. These surface waves exhibited dispersive behavior similar to that observed on metallic spheres [Royer *et al.*, Appl. Phys. Lett. 52, 706-708 (1988)]. [Work supported by the Department of Interior's Bureau of Mines and the Department of Energy.]

11:30

4PA9. Resonant photoacoustic measurements of very low optical absorption in piezoelectric and dielectric crystals. J. D. White, C.

The photoacoustic effect is one of the most sensitive methods for measurements of the low optical absorption in glasses and crystals. Traditional methods using high-powered pulse lasers and attached piezoelectric transducers are limited by noise and scattered light at the transducer. Recently, a new resonant photoacoustic technique has been developed for highly transparent solids, where a cw laser modulated at the acoustic resonant frequency of the sample generates an acoustic signal amplified by the quality factor (Q) of the resonance. This technique is several orders of magnitude more sensitive than conventional pulse techniques. Noncontact capacitive transducers eliminate the problem of scattered light. Two different transduction mechanisms are important in the detection of the acoustic wave at the sample surface. First, the displacement of the surface of the dielectric sample in the electric field of the capacitor results in small changes in the capacitance. Second, through the piezoelectric effect, the strain generates a small electric field at the sample surface; here, the transducer acts as an rf receiver, detecting the changes in the electric field intensity. Based on recent measurements on single crystals of calcium fluoride and quartz, the relative contributions of these two mechanisms will be discussed. [Work supported by the Office of Naval Research.]

4PA10. Measurement and interpretation of laser-generated scan images in anisotropic materials. Wolfgang Sachse (Dept. of Theoret. and Appl. Mech., Cornell Univ., Ithaca, NY 14853) and A. G. Every (Univ. of the Witwatersrand, Johannesburg, South Africa)

This paper describes the results of laser-based *point-source/point-receiver* measurements in single crystal specimens. By combining all the waveforms detected by a pointlike sensor when the source is scanned across the sample, one is able to generate a *scan image*. These images contain the spatial as well as temporal dependence of the ultrasonic signals in the specimen. It is shown that in some cases pronounced anisotropy is observed in the amplitudes which is related to the anisotropy of the material. Further, the signal arrivals can be processed to recover the matrix of elastic constants of the material. Monte Carlo simulations based on ray constructs are used to interpret the scan images. The use of these images as a materials characterization tool is demonstrated. [Work supported by Cornell's Materials Science Center (NSF) and the Office of Naval Research (Physical Acoustics and Solid Mechanics Programs).]

WEDNESDAY MORNING, 1 MAY 1991

INTERNATIONAL C, 8:00 TO 11:30 A.M.

Session 4PP

Psychological and Physiological Acoustics: Perception of Complex Sounds

Marjorie R. Leek, Chair

Army Audiology and Speech Center, Walter Reed Army Medical Center, Washington, DC 20307-5001

Contributed Papers

8:00

4PP1. Simultaneous versus successive intensity comparisons. Huanping Dai and David M. Green (Psychoacoustics Lab., Univ. of Florida, Gainesville, FL 32611)

In profile-analysis experiments, the observer determines whether the amplitude of one or several target components is greater than that of the other nontarget components in a complex. Because the overall level of the stimuli is randomly chosen, detection is based on simultaneous amplitude comparisons of the target and nontarget components. The present experiment examines the ability of observers to make simultaneous versus successive amplitude comparisons of the target and nontarget components. The target and nontarget components were presented either (a) simultaneously or (b) successively in each interval of a 2IFC procedure. In one interval, the amplitude of each component of the target is greater than that of the nontarget; in the other, the amplitude is the same. Three stimulus configurations were used: (1) one target and 20 nontarget components; (2) 10 target and 11 nontarget components; (3) one target and one nontarget component. The threshold was measured as $20 \log(\Delta A/A)$ using an adaptive, maximum-likelihood method tracking 94% correct responses. Results show that the simultaneous comparisons are always 10–20 dB superior to the successive comparisons. [Work supported by NIH.]

8:15

4PP2. Spectral shape discrimination of narrow-band spectra. Bruce G. Berg and David M. Green (Psychoacoustics Lab., Dept. of Psychology, Univ. of Florida, Gainesville, FL 32611)

Previously, it was demonstrated that spectral pitch is used to discriminate three-component, narrow-band spectra when tones are separated by 40 or 80 Hz, but not for a 10-Hz separation. Here, discriminations are investigated based on the temporal envelope. The standard consists of three, equal-intensity sinusoids, with respective frequencies 990, 1000, and 1010 Hz. Presentation level is a random variable with a range of 20 dB and mean 55 dB. Thresholds for an intensity increment of the central tone range from -10 dB to -1 dB *re*: the standard. Thus, performance is as good as that obtained in profile tasks for which components are separated by two or more octaves. Because performance is unaffected by (1) the addition of a modulated masker centered at 4000 Hz [W. A. Yost and S. S. Sheft, *J. Acoust. Soc. Am.* **85**, 848–857 (1989)], or (2) randomization of each component's phase as opposed to the same, fixed phase for each, it does not appear that modulation depth is a critical variable. It is believed that discriminability is related to the spectrum of the envelope. [Work supported by AFOSR, ONR, and NIH.]

4PP3. Dichotic profile analysis. Gail M. Whitelaw, Chien-Yeh Hsu, and Lawrence L. Feth (Div. of Speech and Hearing Sci., Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210)

The majority of profile analysis studies have used monotic and diotic stimulus presentation. Use of dichotic stimulus presentation has been limited in profile analysis experiments, with the presentation configuration primarily consisting of the signal component presented to one ear and the nonsignal components presented to the opposite ear [e.g., Bernstein and Green, *J. Acoust. Soc. Am.* **81**, 1888–1895 (1987)]. In general, results obtained in profile analysis experiments using dichotic stimulus presentation have revealed listener performance inferior to that obtained by either monotic or diotic presentation. Profile analysis is investigated by comparing a monotic condition to a dichotic condition. The monotic signal was generated by producing 21 components using equal logarithmic spacing. Level per component was equal in the standard signal. To create the profile, the odd-numbered components were incremented and the even-numbered components were decremented. The dichotic signal was created by presenting the even-numbered components to the opposite ear. A 2Q-2AFC roving level paradigm was used. Preliminary results suggest that listeners are unable to perform the profile analysis task when information is presented dichotically. Discussion will center on the implications of failing to obtain a dichotic profile.

8:45

4PP4. The roles of pitch differences and modulation incoherence in concurrent sound segregation. Robert P. Carlyon (Exptl. Psychol., Sussex Univ., Brighton BN1 9QG, England), Laurent Demany, and Catherine Semal (Lab. de Psychoacoust., Univ. de Bordeaux, Bordeaux, France)

Listeners discriminated pairs of complex sounds, each consisting of two groups of components. Only those components in the lower-frequency group were resolvable by the peripheral auditory system. For the standard stimulus, the F_0 's of the two groups were frequency modulated coherently at about 125 Hz, so that they were always equal. For the signal, the two F_0 's were modulated incoherently so that they differed by an amount that oscillated between values proportional to the depth of FM. Listeners could perform the discrimination when the zero-peak FM depth was about 7%, and two findings indicate that they did so by simultaneously comparing the pitches of the two groups of components. First, listeners could do the task when the lower group consisted of only odd harmonics of 125 Hz (pitch = 125 Hz), but not when it consisted of even harmonics (pitch = 250 Hz). This shows that they were not comparing the rate of amplitude modulation, caused by beating between adjacent components in the lower group, to that in the upper group: These rates were the same in both conditions. Second, performance was at chance when the F_0 's of the two groups differed (e.g., 100 and 225 Hz), indicating that listeners could not detect FM incoherence *per se*.

9:00

4PP5. Psychoacoustic evaluation of random-like deviation changes in FM signals. Edward Ozimek (Psychoacoust. Lab., Dept. of Psychol., Univ. of Florida, Gainesville, FL 32611)

The detection threshold for frequency modulation was measured at several carrier frequencies. The standard (unmodulated) waveform was a sinusoidal signal of fixed amplitude and fixed frequency. The comparison waveform was frequency modulated. The modulating signal had constant frequency, but the amplitude was a random variable. The amplitude of each successive period was the absolute value of a Gaussian random variable with mean, 0, and standard deviation, σ . The ability of listeners to detect the FM modulation by adaptively varying σ in a two-alternative forced-choice procedure was determined. An increase in the carrier frequency from 120 to 1000 Hz causes, on average, a three-fold increase in the threshold value of σ . Changes in modulation fre-

quencies from 4 to 16 Hz increase the threshold by twofold at the highest carrier frequency. At low-modulation frequencies, different amplitude samples produce statistically different discrimination thresholds. Compared with the classical data for FM signals, the random deviations are easier to hear at higher carrier frequencies and harder to hear at lower carrier frequencies. [Research supported by the Kosciuszko Foundation and the NIH.]

9:15

4PP6. Scalp potentials elicited using AM acoustic signals. William F. Dolphin and David C. Mountain (Dept. of Biomedical Eng., Boston Univ., Boston, MA 02215)

Temporal information is one of the most fundamental aspects of an acoustic signal extracted by the auditory system. Especially as regards communication sounds, temporal patterning is an extremely important information carrying feature of complex signals. Modulation of signal amplitude, at frequencies up to several hundred Hz, is common in many species specific vocalizations as well as in human speech. Scalp potentials which follow the low-frequency envelope of an amplitude-modulated (AM) stimulus waveform were evoked and recorded from anesthetized gerbils. This envelope following response is presumably due to the synchronized discharge of populations of neurons in the auditory pathway. When the carrier frequency (f_c) of an AM constant-frequency acoustical stimulus is greater than the electrical cutoff frequency of the inner hair cells and the modulating frequency (f_{mod}) is much below this cutoff, the instantaneous firing rate of the auditory nerve approximately follows the envelope of the stimulus. As the maximum amplitude of the basilar membrane response occurs at the region corresponding to f_c , it is hypothesized that inner hair cells in this region are preferentially stimulated. To test this hypothesis, AM stimuli in the presence of pure-tone maskers of varying frequencies were presented and scalp potentials recorded. A strong envelope following response, at the frequency of the modulator, was obtained to AM stimuli alone. The response was quite robust and could be elicited using a wide range of f_c 's, modulation frequencies, and stimulus intensities. The response could be eliminated by the simultaneous presentation of a pure-tone masking stimulus (f_{mask}). When f_{mask} was close to f_c , responses at the modulation frequency were diminished by up to 25 dB. Measurements of the group delay, determined from the phase of the response relative to the stimulus phase, indicate at least three separate brain regions as the site of this response. The responses come from progressively more central regions as the modulation frequency is reduced. Such studies may yield valuable insights to the processing of complex stimuli by the auditory system.

9:30

4PP7. The role of envelope cues for the detection of a tone added to a narrow band of noise. Virginia M. Richards (Dept. of Psychol., Univ. of Pennsylvania, 3815 Walnut St., Philadelphia, PA 19104) and Robert D. Nekrich (Univ. of Pennsylvania, Philadelphia, PA 19104)

Using a 2IFC paradigm, psychometric functions for the detectability of a tonal signal added to a 40-Hz-wide band of Gaussian noise was measured using signal and center frequencies of 600, 1800, and 5400 Hz. In a second condition, the noise-alone and the tone-plus-noise stimuli were scaled so as to yield identical energy values across the two intervals of each trial. Under the assumption of optimal combination of independent cues, a comparison of these conditions allows an estimate of the relative contribution of energy-based and non-energy-based cues for the detection of a tone added to noise. In a third condition, the envelopes of the scaled noise-alone and tone-plus-noise stimuli were extracted, and used to modulate tones of either 600, 1800, or 5400 Hz. The ability to discriminate between envelope patterns was measured, and that measure used to estimate each observer's reliance on envelope-based cues. Large individual differences were obtained. For one observer, the detection of a tone added to Gaussian noise was determined to rely equally on envelope-based and energy-based cues. A second observer was found to

depend almost wholly on changes in energy. None of the observers appeared to rely on changes in fine structure that occur when a tone is added to noise, and consistent with that observation, no differential effect of frequency region was obtained. [Work supported by the National Institutes of Health.]

9:45

4PP8. Detection of intensity decrements followed by increments. Stanley Sheft and William A. Yost (Parmly Hearing Inst., Loyola University, 6525 N. Sheridan Rd., Chicago, IL 60626)

Data were collected from a modified decrement detection procedure in order to compare subject performance to the predictions of two decision statistics derived from the output of an envelope detector. At roughly the midpoint of each stimulus presentation, there was an inten-

sity increment of the gated wideband noise carrier. The task was to detect an intensity decrement just preceding the increment in the signal interval of the 2IFC task. Decrement duration ranged from 2.5 to 40 ms. Increments of 3, 6, and 9 dB were used with the onset of the increment randomly occurring 150–300 ms from the stimulus onset. Decrement detection was also measured for conditions in which there was no increment. Best performance was obtained with the 3-dB increment. With the 6- or 9-dB increment, thresholds were significantly higher and showed less change as a function of decrement duration. Simulations based on the output of an envelope detector used either the variance or the ratio of the largest-to-smallest value as the decision statistic. For both statistics, simulations approximated subject performance in the conditions with no increment. Simulations, however, did not show the drop in performance with the 6- or 9-dB increment, suggesting involvement of multiplicative internal noise in envelope detection. [Work supported by NIDCD and AFOSR.]

10:00–10:15

Break

10:15

4PP9. Discrimination of arrhythmic tonal sequences: Effect of delay in onset of second sequence. R. D. Sorkin and D. A. Montgomery (Dept. of Psychol., Univ. of Florida, Gainesville, FL 32611)

These experiments continued tests of the pattern correlation model [R. D. Sorkin, *J. Acoust. Soc. Am.* **87**, 1695–1701 (1990)] of temporal pattern discrimination. Subjects were presented with two sequences (0.6 s) of eight brief marker tones, and had to report whether the sequences had the same or different patterns of marker inter-onset times. The first sequence was presented to the subject's left ear (all marker tones at 1000 Hz) and the second to the right ear (all tones at 2300 Hz). The experimental variable was the delay in the onsets of the two sequences. The delay in sequence onsets varied from 0 to 2.5 s, and was fixed within a block of trials. In some conditions, the marker and gap durations in the second sequence were randomly expanded by a small uniform factor ($0.8 < k < 1.2$). When the sequences were presented nearly simultaneously, temporal manipulation of the second sequence produced large decreases in listener performance. This result is consistent with an envelope correlation mechanism, and differed from the effects observed when the sequences did not overlap in time. In the latter conditions, performance was not sensitive to the expansion manipulation, consistent with the prediction of the pattern correlation model. [Work supported by AFOSR.]

10:30

4PP10. Detecting the temporal asynchrony among components of multicomponent complexes. Jan Zera and David M. Green (Psychoacoust. Lab., Dept. of Psychol., Univ. of Florida, Gainesville, FL 32611)

The listener's task was to detect onset or offset asynchrony among sets of equal-amplitude sinusoids. In the standard condition, all components were started (or stopped) at the same time. In the asynchrony condition, their onset (or offset) times were chosen from a Gaussian

distribution with a certain standard deviation. The value of this standard deviation was adjusted in an adaptive 2AFC procedure to determine the asynchrony threshold. Parameters investigated, in addition to the differences between onset and offset asynchrony, were rise (fall) time of the components, frequency spacing (linear and logarithmic) of the components, and the number of components. The obtained asynchrony thresholds measured in 2AFC tasks ranged from 0.1 to 3 ms for 20-component complexes. In these experiments, the temporal events to be compared occurred in different frequency channels. A control experiment was conducted which indicated that the obtained results could not be explained on the basis of the perception of the wideband waveform.

10:45

4PP11. Human auditory temporal summation in the presence of ipsilateral or contralateral continuous tones. Vishwa K. H. Bhat (Dept. of Commun. Disord., William Paterson College, Wayne, NJ 07470) and George M. Gerken (Callier Ctr. for Commun. Disord., Univ. of Texas, Dallas, TX 75235)

The auditory temporal summation function was measured in 16 normal hearing human subjects using a set of five single-burst and seven multiple-burst tones of 2.0 kHz. In each of two experiments, thresholds were obtained: in the presence of a 60 dB SPL continuous tone one-third octave above or below the stimulus frequency of 2.0 kHz, and in the quiet condition. In experiment I, the continuous tone was presented ipsilaterally and in experiment II, the tone was presented contralateral to the stimulus. It was shown that the ipsilateral tone conditions significantly reduced the slope of the temporal summation functions, whereas the contralateral continuous tone conditions had no influence. Second, it was concluded that a common mechanism processed both single- and multiple-burst stimuli. Finally, it was concluded that a power function was a simple and adequate way of describing the temporal summation process.

4PP12. Temporal effects obtained with signals of various bandwidths presented in notched and unnotched maskers. Beverly A. Wright (Dept. of Psychol., Univ. of Texas, Austin, TX 78712)

The temporal effect, or decrease in simultaneously masked threshold as signal onset is delayed relative to masker onset, was measured for signals centered at 2500 Hz that ranged in width from tonal to 7900 Hz. The masking noises covered the frequency range from 100 to 8000 Hz and either contained a spectral notch that always perfectly complemented the width of the signal, or contained no spectral notch. In the short-delay and long-delay conditions, the 20-ms signal was gated 1 or 250 ms, respectively, after the onset of a 420-ms masker. In the burst condition, the signal was gated 1 ms after the onset of a 23-ms masker. Although there were marked individual differences, with the notched masker, thresholds were as much as 7–17 dB higher in the short-delay condition, and 20–27 dB higher in the burst condition, than in the long-delay condition. With the unnotched masker burst/short-delay differences were minimal. Thus, only the magnitude of the temporal effect obtained with notched maskers was strongly influenced by the choice of reference condition. [Work supported by NIDCD.]

4PP13. Forward masking functions of 3-month-old human infants. G. Cameron Marean and Lynne A. Werner (Dept. of Speech & Hearing Sci., WJ-10, Box 47, Univ. of Washington, Seattle, WA 98195)

Forward masked thresholds for a 1000-Hz tone were measured for 3-month-old infants and for adults. The masker was a broadband noise burst, 100 ms in duration. Masker level was either 65 or 75 dB SPL. The tone had 16-ms rise, 16-ms fall, and no steady state duration. Masked thresholds were measured at Δt of 5, 10, 25, and 200 ms. The Observer-based Psychoacoustic Procedure [Olsho *et al.*, *Devel. Psychol.* **23**, 627–640 (1987)], was used to estimate infant thresholds. Compared to published data from adult listeners, 3-month-olds appear to show relatively more masking at short Δt . In addition, infants may exhibit masking at very long values of Δt , suggesting an additional nonsensory source of masking under certain conditions. Finally, the effects of increasing masker level on the amount of masking seen in infants will be examined. [Work supported by DC00396 to L. A. Werner.]

WEDNESDAY MORNING, 1 MAY 1991

INTERNATIONAL D, 9:00 TO 10:50 A.M.

Session 4SA

Structural Acoustics and Vibration: Active Control of Structural Radiation

Vasundara V. Varadan, Cochair

Department of Engineering Science and Mechanics, Pennsylvania State University, University Park, Pennsylvania 16802

Wayne T. Reader, Cochair

Vector Research Company, Inc., 6903 Rockledge Drive, Suite 1200, Bethesda, Maryland 20817

Chair's Introduction—9:00

Contributed Papers

9:05

4SA1. A wave-number domain approach to the active control of structure-borne sound. Chris R. Fuller and Ricardo A. Burdisso (Mech. Eng., 204 Randolph Hall, Virginia Polytechnic Inst. and State Univ., Blacksburg, VA 24061)

Previous work has shown the possibilities of active control of structurally radiated sound by applying control forces directly to the structure while error information is taken from microphones placed in the radiated far field is impracticable in many applications. On the other hand, it has been demonstrated that simply minimizing the structural response, i.e., by using accelerometers as sensors, may lead to an increase in the radiated sound levels due to "control spillover." Thus, this research work demonstrates for the first time the design of feed-forward controller in the wave-number domain for eliminating the use of microphones in the radiated far field. By using a stationary phase approach to evaluate sound radiation from panels, it is shown that only one spectral wave-number component of the structural response radiates towards one particular angle. Thus, by minimizing particular wave-number components on the structure, the sound radiation can be minimized at the corresponding radiation angles. A 2-D baffled simply supported beam

example demonstrates this concept. [Work supported by ONR/DARPA.]

9:20

4SA2. Application of surface waves for antifouling. B. Shanker, V. V. Varadan, V. K. Varadan, and B. A. Shaw (Dept. of Eng. Sci. and Mech., Res. Ctr. for Eng. of Electron. and Acoust. Mater., Penn State Univ., 227 Hammond Bldg., University Park, PA 16802)

Marine fouling in ships and biofouling in condenser tubes, etc. cause numerous problems and a large reduction of fuel efficiency. Conventional methods include toxic paints and hull vibration. A nontoxic solution would be to excite Love waves on the surface and prevent the very first bio-attachment on the surface. A thin PZT film with IDTs (InterDigital Transducers) flush mounted to the surface is used to launch these waves. Experimental results of this technique, which seems viable and cost effective, and most importantly, nontoxic, will be discussed.

4SA3. Active control of sound radiation from a uniform rectangular fluid-loaded plate. Yi Gu and Chris R. Fuller (Dept. of Mech. Eng., Virginia Polytech. Inst. and State Univ., Blacksburg, VA 24061)

Active control of sound radiation from a simply supported rectangular one-side fluid-loaded plate is analytically studied. The plate is assumed to be excited by a point force at subsonic frequencies. The dynamic equation of the plate motion is based on the *in vacuo* eigenfunctions of a homogeneous panel as the basis for Fourier decomposition of the fluid loading. Feed-forward control is applied by either point forces or piezoelectric actuators attached to the plate. The amplitudes of the control forces are determined by the optimal solution of a quadratic cost function which integrates the far-field radiated acoustic pressure in a hemisphere in the fluid half-space. The results show that for on-resonance frequencies, high global reduction in radiated pressure is possible with two active forces properly located on the plate. For off-resonance conditions in order to provide global control an increased number of control forces is required. However, for the low frequencies considered here four control forces proved adequate. The far-field directivity pattern, the modal amplitudes of the plate vibration, the plate power spectrum in a two-dimensional wave-number domain, and the near-field intensity distribution are extensively studied. The work confirms that the feed-forward control technique will work for heavy fluid loading where the plate modes are nonorthogonal and mode edge coupling is important. [Work supported by ONR/DARPA.]

9:50

4SA4. "Smart" noise control foams: A feasibility study. J. S. Bolton, N.-M. Shiau, and E. R. Green (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907)

A porous layer can absorb a significant amount of acoustical energy only if its thickness is comparable to the wavelength of the incident sound. Thus, a porous layer inevitably becomes a less effective sound absorber as the frequency is decreased. In this paper it will be shown through theoretical calculations that the low-frequency performance of a finite-depth layer of elastic porous material may be enhanced by applying an appropriate force to the solid phase at the front surface of the layer. In particular, it will be shown that at any angle of incidence the solid phase may be forced in such a way to create a perfect impedance match with an incident plane wave, thus causing the sound to be completely absorbed. Complete absorption is not possible when the incident sound field is diffuse; nevertheless, sample calculations will show that considerable enhancement of the passive absorption is still possible. Note that the success of the approach suggested here requires a significant degree of coupling between the motion of the solid and fluid phases of the porous material. Thus it may be expected that partially reticulated, polyurethane foams will be susceptible to this approach owing to the degree of viscous and inertial coupling between their fluid and solid phases. Several methods for actuating the front surface of an elastic porous material and for sensing its surface normal impedance will be discussed.

10:05

4SA5. Numerical simulation of active noise control for three-dimensional structures: Preliminary investigation. Christopher E. Ruckman and Christopher R. Fuller (Dept. of Mech. Eng., Virginia Polytech. Inst. and State Univ., Blacksburg, VA 24061)

In certain cases, low-frequency noise radiated by a structure into a surrounding fluid can be attenuated by applying harmonic "control forces" directly to the structure. If the system is controllable, i.e., if the proposed control forces are sufficiently capable of influencing the radiation, then global radiation attenuation is possible. Recent studies show that some structures, including baffled flat plates, are indeed controllable in this sense. But analyzing a complex three-dimensional structure,

particularly if it is immersed in a dense fluid, can be difficult because there usually is no analytical solution for the radiated field. The present research adapts NASHUA, a general finite element/boundary element computer program, to investigate whether radiation from three-dimensional structures can be controlled by harmonic forces. The preliminary test case to be presented is a thin spherical shell, chosen because it has an analytical solution. Control force locations are determined by examining the spatial Fourier transform of the surface velocity. Control force magnitudes and phases are found by minimizing the radiated power using quadratic optimal control theory. Also examined are sensitivity to control-force placement and phasing errors, and sensitivity to system identification errors. The numerical and analytical results are found to be in good agreement. [Work supported by ONR/DARPA.]

10:20

4SA6. Digital time delay network for active underwater acoustic coating. Thomas R. Howarth (HVS Technologies, 820 N. University Dr., State College, PA 16803), Xiaoqi Bao, Vijay K. Varadan, and Vasundara V. Varadan (Dept. of Eng. Sci. and Mech. Res. Ctr. for the Eng. of Electron. and Acoust. Mater., Penn State Univ., 149 Hammond Bldg., University Park, PA 16802)

A digital time delay system has recently been developed for control of an active underwater coating. The active coating contains piezoelectric polymer sensors and a piezocomposite actuator encapsulated within a host polymer. The coating is for layering on submerged objects to actively reduce underwater sound reflection. The control system incorporates time delay circuitry as opposed to the analog phase delays used in previous studies. The type of control is an open-loop velocity feedback. This system prevents feedback instability because the acoustic reflection is decoupled from the actuator. The digital controller can characterize both the incident acoustic pressure and the reflected pressure through parallel multichannel processing. The reflected pressure is monitored at the sensors for determination of the effective sound reduction. Experimental results were obtained in a water-filled acoustic pulse tube terminated at its boundary with the active coating. Echo reduction comparisons as functions of sound reduction and bandwidth will be presented.

10:35

4SA7. Active control of edge sound radiation from a clamped beam. Catherine Guigou and Chris R. Fuller (Dept. of Mech. Eng., Virginia Polytech. Inst. State Univ., Blacksburg, VA 24061)

Active control of sound radiation from a thin semi-infinite beam clamped at one end is analytically studied. Active control is achieved by applying either control forces (approximating shakers) or control moments (approximating piezoelectric actuators) on the beam near the edge discontinuity. For a single frequency, the flexural response of the beam subject to an exciting point force and to the control forces or control moments is expressed in terms of waves of both propagating and near-field types. The control force or control moment magnitudes are derived by optimizing a quadratic cost function which is defined as the acoustic power radiated from one side of the baffled beam and obtained by integrating the far-field radiated acoustic intensity over a hemisphere. For single frequencies, large attenuation in radiated acoustic power and pressure is found when one and two control forces or two and four paired control moments are applied. The proper location of the control forces or control moments is determined in order to obtain the maximum decrease of the optimized radiated acoustic power. The influence of the control near-field component on the amount of attainable attenuation is studied and found to be important. The far-field radiated pressure directivity, the displacement of the vibrating beam, and the one-dimensional wave-number spectrum of the beam velocity are extensively studied. The work provides insight into active control of edge radiation from more complex 2-D structures. [Work supported by ONR/DARPA.]

Session 4SP

Speech Communication: Segmental Acoustic Properties

Mark Liberman, Chair

Linguistics Department, University of Pennsylvania, Williams Hall, Philadelphia, Pennsylvania 19147

Contributed Papers

8:00

4SP1. Velar height and sentence length: Declination? F. Bell-Berti (St. John's Univ., Jamaica, NY 11439 and Haskins Labs., New Haven, CT 06511) and Rena A. Krakow (Temple Univ., Philadelphia, PA 19122 and Haskins Labs., New Haven, CT 06511)

Considerable evidence attests to the phenomenon of F_0 declination during the course of an utterance. Recently, the question of whether upper articulators might show declination-type effects has been raised for the jaw [E. V. Bateson and C. A. Fowler, *J. Acoust. Soc. Am. Suppl.* 1 **84**, S128 (1988)]. In previous studies, lower velar peaks were noted in sentence-final than -initial syllables, prompting a systematic investigation of velar declination. Thus peak velar height for early and late syllables in natural and reitarent-speech sentences of from three to nine syllables has been examined. Velar position and acoustic data were collected from three speakers of American English. Differences in velar height were examined with respect to number of syllables and acoustic duration. Possible interactions between declination-type processes and factors such as stress were also studied, since stress has been shown to have a strong influence on velar movement [R. A. Krakow, *J. Acoust. Soc. Am. Suppl.* 1 **82**, S17 (1987); J. Vassiere, *Phonetica* **45**, 122-139 (1988)]. [Work supported by NIH Grants DC-00121 and RR-05596 to Haskins Labs.]

8:15

4SP2. The F_3 trajectories of American English /r/'s. Suzanne Boyce and Carol Y. Espy-Wilson (Res. Lab. of Electron., MIT, Cambridge, MA 02139)

It is well known that the coarticulatory influence of /r/ on preceding segments, as evidenced by a lower than expected F_3 frequency, may be substantial. Researchers who have studied this phenomenon have found that this "spreading" occurs under a number of different conditions (cf. Espy-Wilson, 1987). A unifying explanation may be that the articulatory gesture for /r/ has a relatively incompressible trajectory whose timing "slides" with respect to other gestures in a sequence. This hypothesis was tested acoustically by analyzing the time course of the F_3 trajectory in several American English speakers' production of the words "varam," "vavram," and "vavam." In these data, previous findings of "spreading" to the preceding segments were confirmed. In addition, the /r/'s in "varam" and "vavram" showed a nearly identical time course. As a consequence, F_3 during the post-vocalic /v/ of "vavram" is considerably lower than was seen in the post-vocalic /v/ of "vavam." This suggests that the explanation for spreading of the feature retroflex into the /v/ is due to a constrained time course for /r/. This hypothesis will be further explored in words when /r/ cooccurs with /g/, a segment whose tongue-body requirement potentially conflicts with the articulatory requirement for /r/.

8:30

4SP3. Acoustic parameters for place of articulation in nasals and stops. Kathleen Mary Kurowski (Dept. of Cognitive and Linguistic Sci., Box 1978, Brown Univ., Providence, RI 02912)

A metric that identified labial and coronal place of articulation in English nasal consonants with over 80% accuracy had been developed by correlating distinctive patterns of rapid energy change at the nasal release with each place of articulation [K. M. Kurowski and S. E. Blumstein, *J. Acoust. Soc. Am.* **81**, 1917-1927 (1987); K. M. Kurowski, unpublished dissertation, Brown Univ. (1990)]. This study attempted to address two questions. (1) Could the same metric be successfully applied across manner to classify place of articulation in homorganic voiced stops? (2) In the event that the metric failed here, to what extent did the burst onset alter the patterns of rapid energy change predicted by the metric? The same three speakers used in the nasal study (1990) produced CV syllables, consisting of [b,d] followed by [i,e,a,o,u]. The results showed that the classification scores across speakers for the voiced stops (labial = 64%; coronal = 83%) were lower than those for nasals (labial = 79%; coronal = 87%). Still, the general patterning of the results was the same across manner: Labial scores remained lower than those of the coronals due to the intractable problem of classifying labials in the environment of [i]. Two reanalyses using different window sizes demonstrated that the aperiodicity of the initial portion of the burst inflated some targeted areas of energy change, adversely affecting the performance of the metric. Various approaches to the question of how to reconcile formal aspects of the present metric with the added complexity introduced by the voiced stops are discussed in the light of their ramifications for the theory of acoustic invariance. [Work supported by NIH]

8:45

4SP4. An acoustic phonetic description of the American English /ð/ in two modes. Josephine Horna (Dept. of Linguistics/ISALS, New York Univ., 719 Broadway, New York, NY 10003)

Phoneticians traditionally describe American English /ð/ as a voiced, nonsibilant, interdental fricative. However, in flowing speech, /ð/ often sounds stoplike. This study investigates the acoustic attributes of stoplike, as well as continuant, variants of /ð/ in "the" occurring in various phonological conditions in text-reading and conversation modes. The results of spectrographic analysis allow for the definition of five degrees of constriction and two types of constriction release, burst and smooth formant transition into the following segment. The degree of constriction is a coarticulatory effect of the preceding phoneme. Smooth formant transitions tend to occur with less constricted degrees of constriction; a burst may occur with any degree of constriction and is prevalent across phonological conditions. Articulatory interpretations that explain these and other variations of /ð/ are forwarded. The results also show intermodal differences in /ð/ in that there is a higher

occurrence of some coarticulatory effects of the preceding phoneme in the conversation mode than in the text-reading mode.

9:00

4SP5. Voiced versus voiceless inaspirates: Acoustic measurements of English stops. Laura L. Koenig, Arthur S. Abramson, and Leigh Lisker (Haskins Labs., 270 Crown St., New Haven, CT 06511-6695)

Attention is focused on intervocalic stops in trochaic words. Embedded in semantically plausible sentences, tokens of six pairs of words like *robing* and *roping* were recorded by three speakers of American English. These included only places of articulation, labial and dorso-velar, for which the voicing opposition is reliable for all dialects in natural speech. Four acoustic properties were examined because they are considered relevant and they are amenable to editing for perceptual experiments. They are closure pulsing, closure duration, stressed-vowel duration, and release-burst amplitude. The only entirely reliable correlate is presence versus absence of closure pulsing. A fairly good correlate is vowel duration. Vowels before voiced stops are somewhat longer, although there is overlap between the two ranges. Another fairly good correlate is closure duration. Voiceless closures are somewhat longer, although here, too, there is overlap. The amplitude of the burst is a poor correlate, even though voiceless bursts tend to be more intense. These data furnished a baseline for our paper on perception. [Work supported by NIH Grant HD-01994.]

9:15

4SP6. Voiced versus voiceless inaspirates: The perception of English stops. Leigh Lisker, Arthur S. Abramson, and Laura L. Koenig (Haskins Labs., 270 Crown St., New Haven, CT 06511-6695)

Preliminary data on four acoustic variables as possible cues to the contrast between voiced and voiceless unaspirated stops in fluent American English speech were reported at the 119th Meeting. They were

burst amplitude, duration of the preclosure vowel, duration of the closure interval, and the nature of the closure signal. A single English speaker recorded the sentence pair *He's known as Jack the Ribber* and *He's known as Jack the Ripper*, which served as sources of waveform-edited stimuli that were identified by a jury of ten American English speakers. Additional word identification data have since been collected for three additional sentence pairs recorded by the same single speaker, with the new word targets chosen to include dorsal as well as labial stops after both the short vowel [ɪ] and the long vowel [o]. For each acoustic variable two values were tested, one normal for the word as originally produced and the other typical of the minimally contrasting word. The data allow us to rank the four acoustic properties in respect to their roles in perception: burst amplitude < vowel duration < closure duration < closure signal. [Work supported by NIH Grant HD-01994 to Haskins Labs.]

9:30

4SP7. The production of "breathy voice." Katherine Davis (Dept. of Linguistics, Morrill Hall, Cornell Univ., Ithaca, NY 14853)

The nature of voicing in the Hindi "voiced aspirated" phonemes has been a subject of discussion in the literature for a number of years. In particular, the question of whether or not "breathy voice" or "murmur" is present during the aspirated portion of the stop has been at issue. The present study looks at productions of the velar voiced aspirated stop (/gʱ/) in initial position from the speech of ten adult female speakers of Hindi. Amounts of lead time (prevoicing) and lag time (post-release VOT) in the voiced aspirate were compared to lead and lag times in the voiced unaspirate (/g/) and the voiceless aspirate (/kʰ/). In addition, the presence or absence of voicing during the post-release portion of the stop was determined. Both lead and lag time values of the voiced aspirate were found to be significantly different from those of the other velar stop types, and the presence of voicing during aspiration (breathy voice) was found to be speaker dependent. Implications of these findings for a description of voicing contrasts in various languages will be discussed.

9:45-10:00

Break

10:00

4SP8. Production and perception of vowel duration as a cue to the word-final English /t/-/d/ contrast by native and Chinese subjects. J. E. Flege (Univ. of Alabama at Birmingham, Box 503, UAB Station, Birmingham, AL 35294)

Vowel duration was measured in seven minimally paired English words such as *beat-bead* and *bat-bad* as spoken by native English (NE) subjects and four groups of Chinese subjects (ten per group). The Chinese subjects were either relatively inexperienced (MA, TA) or experienced (TB) adult learners of English, or else had learned English as an L2 before the age of 12 years (TC). The early learners in TC closely resembled the NE speakers in making vowels much longer before /d/ than /t/. The inexperienced adult learners (MA, TA) produced a voicing effect, but one that was far smaller than that of the subjects in NE and TC. The experienced adult learners in TB produced a significantly larger voicing effect than the inexperienced subjects in TA and MA, but their effect was also smaller than that of the subjects in NE and TC. Sensitivity to vowel duration as a perceptual cue to the English /t/-/d/ contrast was investigated in three ways. The subjects identified the

members of natural continua in which vowel duration was varied by deleting glottal pulses. One continuum ranged from *bead* to *beat*, the other from *bad* to *bat*. The subjects imitated the members of both continua, and also chose which members of the continua represented the best exemplars of *bead*, *beat*, *bad*, and *bat*. As expected from the production data, the early learners in TC resembled the NE speakers perceptually, whereas the inexperienced adult learners (MA, TA) showed little evidence of perceptual sensitivity to vowel duration. Some experienced adult learners in group TB, on the other hand, were perceptually sensitive to the vowel duration cue. The relationship between their production and perception will be discussed.

10:15

4SP9. VOT values in bilinguals versus monolinguals of Hindi and English. Carol E. Carey, H. S. Gopal, and Heidi Affentranger (Dept. of Speech and Hearing Sci., Univ. of California, Santa Barbara, CA 93106)

This study compared the VOT values of stop consonants produced

by bilingual speakers of Hindi-English with those produced by monolingual speakers of Hindi and English. Specifically, it investigated whether native Hindi speakers use their native VOT values of stop consonants when producing English stops, or if they use the correct/appropriate VOT values as produced by native English speakers. Two subjects in each of the three speaker groups (Hindi monolingual, English monolingual, and Hindi-English bilingual speakers) produced CVC test syllables in a sentence context. These sentences were acoustically analyzed for the VOT values of the initial stop consonants in the CVC syllables. Results will be discussed within the framework of the equivalence classification hypothesis which claims that, when there are similar phones between two languages, bilingual speakers of those languages will not produce the phones in the second language accurately as they will substitute the similar phone from their native language [J. Flege, *J. Phon.* 15, 47-65 (1987)].

10:30

4SP10. Factors influencing the spectral representation of front-back vowels in American English. H. S. Gopal, Joyce Manzella, and Carol Carey (Dept. of Speech and Hearing Sci., Univ. of California, Santa Barbara, CA 93106)

It has been proposed that front vowels in American English have an auditory distance of less than 3 critical bands between the second and third formants, whereas, in back vowels, this distance is greater than 3 Barks [A. K. Syrdal and H. S. Gopal, *J. Acoust. Soc. Am.* 79, 1086-1100 (1986)]. The current study investigates the success of classifying front and back American English vowels into two separate classes using this auditory model given the variations in the acoustic signal due to speaking rate, post-vocalic consonantal voicing, and different speakers. Several researchers have shown that these factors influence vowel formant frequencies. Measurements were made of vowel F_1 , F_2 and durations in /pVC/ target syllables containing one of four pairs of front-back vowels and voiced or voiceless stops or fricatives. Target syllables were produced in sentence context at three different speaking rates by two male and two female native American English speakers. Results will be reported and discussed for an invariant auditory representation of front and back vowels in AE under these influences.

10:45

4SP11. A comparison of the first and second formants of vowels common to English and French. Dawn M. Behne (Speech Res. Lab., Dept. of Psychol., Indiana Univ., Bloomington, IN 47405)

English vowels have been characterized as being lower, more central, and less rounded than the predominantly high, front, rounded vowels of French [P. Delattre, *Comparing the Phonic Features of English, French, German and Spanish: An Interim Report* (Groos Verlag, Heidelberg, 1965)]. In the present study, Delattre's description is experimentally investigated by comparing the first and second formant frequencies of English and French vowels that occur in both languages (i.e., /i, e, ε, u, o, ɔ, ɒ/). The results suggest that: (1) high vowels tend to be higher in French than in English; (2) /a/ tends to be lower in English than in French; (3) high and mid vowels tend to be more central in English than in French; (4) rounded vowels tend to have greater liprounding in French than in English; (5) front vowels are higher and more fronted in French than in English; and (6) back vowels have greater pharyngeal constriction in English than in French. Although Delattre's description of English and French vowels is generally supported by the results, it does not fully characterize the differences demonstrated between the vowels common to English and French.

11:00

4SP12. Formant frequencies of lexically unstressed English vowels in two speech modes. Karen L. Wallace (Dept. of Linguistics/ISALS,

New York Univ., 719 Broadway, New York, NY 10003)

Most English unstressed vowels are perceived as varying between schwa and /ɪ/. To investigate this variation, which can be only partially explained by coarticulation with consonantal contexts, words containing such vowels in a wide variety of C _ C environments were randomly selected from the conversation of a male speaker of American English. These were compared to the same words in the speaker's reading of sentences from his conversation. Frequency levels of the first two formants were measured at three points within each vowel. A statistically significant percentage of word pairs demonstrated larger F_2/F_1 ratios in the sentence-reading mode, particularly for formant values obtained at vowel midpoints. Because this result occurred with all consonantal contexts, it can be concluded that unstressed vowels are, in general, more /ɪ/-like in formal speech. Examination of the word pairs that did not conform to this predominant pattern revealed unexpected correlations between formant frequencies of an unstressed vowel and two higher level linguistic factors: position in the word and proximity to a primary stress.

11:15

4SP13. Effects of speaking rate on the vowel length distinction in Korean. Harriet S. Magen and Sheila E. Blumstein (Dept. of Cognitive and Linguistic Sci., Brown Univ., Box 1978, Providence, RI 02912)

Variations in speaking rate affect the perception of temporally defined phonetic distinctions in consonants, such as voicing in stops, and the stop-glide and stop-affricate contrasts. The production data on these contrasts show asymmetrical effects on temporally distinctive pairs, with the longer members showing the effects more markedly. These data further suggest that the shorter member of the pair serves as a phonetic anchor. This study investigates whether similar effects emerge for vowels. To this end, production of short-long vowel pairs was examined across speaking rates in Korean, a language that has phonemic vowel length. Results show that both long and short vowels vary across speaking rates, such that the symmetrical effects found for consonants are not found for vowels. Short vowels do not provide a phonetic anchor. In fact, the durations of short vowels produced at a slow rate nearly always overlap those of long vowels produced at a fast rate. [Work supported by NIH.]

11:30

4SP14. Vowel effects of pharyngeal and uvular consonants in Interior Salish. Nicola J. Bessell (Dept. of Linguistics, Univ. of British Columbia, 4616 Larchwood Ave., Philadelphia, PA 19143)

Formant values of the vowels /i/, /u/, /a/, and schwa in the environment of the pharyngeal and uvular consonants of several Interior Salish languages are plotted and the results are compared with the formant values of the same vowel phonemes in nonuvular/pharyngeal environments. The data represent several speakers of each language. Results show a consistent lowering of vowels in uvular/pharyngeal environments, with F_1 rising considerably. The rise in F_1 is greatest for vowels in pharyngeal environments. Backing (or F_2 lowering) is not consistent across vowels or place of articulation. In fact, the voiceless pharyngeal shows some tendency to front the low central vowel /a/. The data are compared to similar material from several Arabic dialects.

11:45

4SP15. Study of vowel coarticulation in British English. James L. Hieronymus (Ctr. for Speech Technol. Res., Edinburgh Univ., 80 South Bridge, Edinburgh, Scotland)

Coarticulation in continuous speech causes the vowel formant tracks to be altered by nearby phonemes. The present study concentrates on 660 phonetically hand-labeled read sentences from one male talker of the RP accent of British English. The 14 monothongal vowels of RP

English, /ii, i, a, e, aa, uh, oo, o, u, u", uu, @@, @, / were studied using formants, duration, and syllable stress as parameters. The formant frequency values near at the right and left edges and the center of the hand-labeled vowel region were studied using scatter plots and statistical analysis. No simple relationship between adjacent phoneme place of articulation and the vowel target change has been found because of the presence of "robust vowels" within each vowel category which always reach their target even in the presence of adjacent semivowels. These vowels are not merely stressed, or in content words; instead, they depend on other factors that are being studied. The long duration (prepausal lengthened) vowels were always robust, but some quite short vowels were also. [Work supported by the Alvey Project.]

12:00

4SP16. The influence of consonant duration on vowel-to-vowel coarticulation in Korean. John-Dongwook Choi (Dept. of Linguistics, UCLA, Los Angeles, CA 90024-1543)

This study examines the influence of durational differences associated with the three manner types (unaspirated, aspirated, fortis) on the coarticulatory effects of one vowel on another across an intervening consonant in Korean. F_2 and consonant durations were measured in V_1CV_2 tokens read by five speakers, where $V_n = \{/i/, /a/\}$ and $C = \{(\text{bilabial, dentoalveolar, palatoalveolar, velar}) \times (\text{aspirated, unaspirated, fortis})\}$. Repeated measures ANOVA tests show that Korean exhibits both anticipatory and carryover vowel-to-vowel coarticulation. There is also anticipatory coarticulation associated with the consonant's manner type, confined predominantly to /a/. The data show greater magnitude differences on $V1 = /a/$ when the consonant is unaspirated (224 Hz) than aspirated (143 Hz) or fortis (127 Hz). There is no statistically significant coarticulation between manner type and $V2$. Durationally, unaspirated C's are approximately half as long as the aspirated and fortis C's. The aspirated and fortis C's are durationally distinguished by differences in VOT. These observations will be discussed in the context of dual channel and single channel models of consonant and vowel production.

WEDNESDAY MORNING, 1 MAY 1991

WASHINGTON, 9:00 A.M.

Meeting of Accredited Standards Committee S2 on Mechanical Shock and Vibration

to be held jointly with the

U.S. Technical Advisory Group (TAG) Meeting for ISO/TC 108 Mechanical Vibration and Shock

S. I. Hayek, Chairman S2

Applied Research Laboratory, Penn State University, P.O. Box 30, State College, Pennsylvania 16801

D. F. Muster, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108
4615 O'Meara Drive, Houston, Texas 77035

Standards Committee S2 on Mechanical Shock and Vibration. Working Group chairs will present reports of their recent progress on writing and processing various shock and vibration standards. There will be a report on the interface of S2 activities with those of ISO/TC 108 (the Technical Advisory Group for ISO/TC 108 consists of members of S2, S3, and other persons not necessarily members of those committees) including a report on the current activities of ISO/TC 108, and plans for the next meeting, to take place in Kobe, Japan from 4-13 September 1991.

Session 5AA

Architectural Acoustics, Noise and Structural Acoustics and Vibration: Ground-Borne Transportation Noise in Buildings

Timothy Foulkes, Chair

*Cavanaugh Tocci Associates, Inc., 327 F Boston Post Road, Sudbury, Massachusetts 01776**Invited Papers*

1:30

5AA1. Characterization of the dynamic response of buildings to ground-borne vibration. Richard A. Carman (Wilson, Ihrig & Associates, Inc., 5776 Broadway, Oakland, CA 94618)

A transfer mobility measurement procedure is proposed as a basis of characterizing the dynamic response of existing buildings to new ground-borne vibration sources. To demonstrate this technique, an empirical model was derived that can be used to predict the response of buildings to rail transit vibration. The proposed testing procedure and model are based on the comprehensive impact-testing procedure proposed by Nelson and Saurenman [Transportation Research Record 1143 (1987)] for predicting rail transportation ground-borne noise and vibration. The transit model for buildings is based on an analysis of measured transfer mobility for residential buildings using an instrumented impact device and an analytic comparison with actual train-induced vibration. Coupling loss between the ground and building foundation is incorporated in the model, as is amplification of building vibration by floor resonances. Examination of resonance conditions is made possible by modal testing. By making adjustments to the model, it is possible to extend the result to other sources, such as roadways.

2:00

5AA2. Traffic-induced vibration study on residential structures in suburban Washington, DC. Scott Harvey and Bob Capozello (Polysonics, Inc., 5115 MacArthur Blvd., NW, Washington, DC 20016)

Based on residents' complaints of traffic-induced vibrations in single family dwelling units, vibration measurements were evaluated in terms of annoyance and structural damage. The study included 15 residences and measurements were made under a variety of traffic conditions. Measurement points within each residence included the foundation and utility pipes for determination of transmission path. Real time, 1/3-oct measurements were made in the range of 0.8–80 Hz and compared with criterion set forth by the Federal Highway Administration. It was found that all the residences were on local bus routes and that buses were the major cause of structural vibration. It was shown that vehicle suspension characteristics had greater influence on structural vibrations than gross vehicle weight. Pavement smoothness was the one controllable factor capable of reducing traffic-induced vibrations. Recommendations were made for decreasing the vibrations to a level lower than the threshold of annoyance.

2:30

5AA3. Effects of track fixation on ground-borne noise of train passage. James D. Barnes and Frank N. Iacovino (Acentech Inc., 125 Cambridge Park Dr., Cambridge, MA 02140)

Vibration measurements were obtained at selected sites along the new rapid transit track of the South-west Corridor in Boston, MA during dedicated rapid transit train passages. The sites were selected to compare the vibrations associated with the following track fixation techniques: direct fixation, floating slabs, direct fixation with Cologne Egg, timber ties on stone ballast, concrete ties on stone ballast, and timber ties in concrete. In addition, ground vibrations associated with revenue commuter and Amtrak train passages were measured at eight community sites prior to and following the realignment and upgrade of this corridor. The various track fixation techniques and measurements will be described and compared for both the rapid transit and commuter/Amtrak service lines. The ground-borne noise expected for each condition will be discussed and compared to appropriate criteria.

3:00

5AA4. Sound isolation design for cinemas below the roadway—A case study. Timothy Foulkes (Cavanaugh Tocci Associates, Inc., 327 F Boston Post Road, Sudbury, MA 01776)

The recently completed Tower City Cinema complex in downtown Cleveland is located below street level. Some of the 11 theaters are below an office building, and several theaters are below an active street. Special construction was required to provide an acceptable theater environment. During design, two of the theaters were designated for THX certification requiring lower background noise than normal cinemas. All theaters achieved a background noise level of NC 30 with no discernible sound or vibration from the traffic. This paper will include preconstruction measurements and analysis, construction details, and post-completion sound measurements.

3:15

5AA5. Estimating interior sound-pressure level arising from ground-borne noise. Douglas H. Bell (Cavanaugh Tocci Associates, Inc., 327 F Boston Post Road, Sudbury, MA 01776)

Ground-borne noise produced by railroads, subways, and roadways can be radiated as airborne noise in enclosed reverberant spaces. A simple model utilizing site vibration measurements to estimate resultant sound-pressure levels is presented. The model is then extended to devise a broad class vibration criteria that can be used to assess the severity of existing ground-borne noise. To demonstrate this model, sound and vibration measurements were simultaneously measured in a basement space located near a subway line. Predicted sound levels derived from the measured vibration are compared with measured sound levels.

3:30

5AA6. The effect of coincidence transmission on field measurement of facade sound isolation. Angelo J. Campanella (Campanella Associates, 3201 Ridgewood Dr., Columbus, OH 43026)

Windows, walls, and doors suffer coincidence transmission (CT).

Sound penetration increases with incidence angle, to a maximum near grazing incidence. Construction material "standard" sound isolation is cataloged from laboratory (ASTM E 90 or ISO 140/III) diffuse incident sound transmission loss (TL) tests. Field facade measurements per ASTM E 966 [J. Acoust. Soc. Am. Suppl. 1 88, S135 (1990)] apply $10 \log(\cos \theta)$ angle correction (proper for open apertures, e.g., ventilators) to compute inside-outside transmission loss (OITL), which is comparable with diffuse TL values. For panels with CT, comparable values using unidirectional (loudspeaker) sound require 45- to 60-deg incidence or "field transmission loss" adjustment CT panel transmissivity derived by Cremer, Beranek, Ver, and Holmer indicates inverse cosine-squared behavior and a transparency ("coincidence") frequency for each angle. This can be integrated over all diffuse angles for comparison to TL values [e.g., R. E. Jones, J. Acoust. Soc. Am. 66, 148-164 (1979); TL = 26-28 dB at 500 Hz for 16-mm (5/8-in.) drywall]. Comparison indicates integration to 78 or 80 deg. Theory/OITL/TL agreement ("match") occurs for unidirectional sound incident at around 60 deg. Theory/TL or OITL/TL data by others for a drywall stud wall, block wall, and a window showed "match" angles from 45 to 60 deg. More CT test data on thin panels and three-way comparisons (theory/OITL/TL) are needed to improve OITL precision and reduce bias while expanding the range of test angles and source types (e.g., traffic).

3:45

5AA7. Sound emission from residential ventilation fans. J. D. Quirt (Acoustics Sec., Inst. for Res. in Construction, Natl. Res. Council Canada, Ottawa, Ontario K1A 0R6, Canada)

This paper describes the acoustics part of a study to evaluate a draft Canadian standard for rating the air-handling and sound emission of residential ventilation fans. This included laboratory sound-power testing of 11 ventilation units using the CSA C260 draft procedure, which combines requirements from ANSI S1.32 sound-power test with installation requirements from the Home Ventilating Institute (industry) test method. The laboratory tests were structured to verify that the method is practicable and to estimate reproducibility of the test method. Subsequent field measurements (on the same fans, when installed in homes) were included to demonstrate the relationship between sound-power ratings and the resulting sound pressure in practical application.

Session 5EA

Engineering Acoustics: Signal Processing and Waves in Elastic Media

Roger T. Richards, Cochair

Naval Underwater Systems Center, New London, Connecticut 06320-5594

Thomas R. Howarth, Cochair

HVS Technologies, Inc., 820 North University Drive, State College, Pennsylvania 16803

Chair's Introduction—1:00

Contributed Papers

1:05

5EA1. Passive fetal heart rate monitor using piezopolymer pressure sensors. Allan J. Zuckerwar (NASA Langley Res. Ctr., M.S. 238, Hampton, VA 23665), John W. Stoughton, Robert A. Pretlow (Old Dominion University, Norfolk, VA 23529-0246), and Donald A. Baker (Holy Family Medical Clinic)

Fetal heart rate monitoring is performed routinely by pulsed Doppler ultrasound, but is currently confined to a clinical setting because of the bulkiness and high cost of the equipment. A passive monitor, using piezopolymer pressure sensors located on a belt worn by the mother, has been developed to detect the fetal heart tone and to determine the fetal heart rate through appropriate signal processing. The sensors are constructed to fulfill the following functions: detecting pressure pulses on the abdominal surface, shielding against 60-Hz interference, isolating the patient electrically, insulating against ambient sound, and canceling maternal rigid-body motion. The signal processing employs linear prediction techniques to identify the fetal heart tone from among competing background signals and yields a real-time evaluation of the fetal heart rate. The monitor has been applied to conduct clinical fetal nonstress tests simultaneously with ultrasound, and the results are compared. The monitor is not only inexpensive but lends itself to an ambulatory mode of operation, whereby the mother can conduct the fetal nonstress test in her home.

1:20

5EA2. Signal processing for an acoustically based fetal heart rate monitor. Robert A. Pretlow and John W. Stoughton (Dept. of Electr. and Computer Eng., Old Dominion University, Norfolk, VA 23529-0246)

A least-mean-square (lms) linear prediction algorithm has been developed to accomplish detection of fetal heart tones and thereby derive heart rate from a raw signal generated by a previously described passive acoustical sensor array. The desired heart tone signal has a characteristic signature but is of extremely low amplitude and contaminated with noise consisting of large-amplitude maternal heart tones, abdominal sounds, body motion, and environmental sounds, and also mild 60 Hz. The predictor coefficients were derived by adaptively "training" on ideal fetal heart tones recorded from several patients. The lms algorithm detects a heart tone event when the predictor mean-square error falls below an adaptively updated threshold level. The algorithm contains logic for correction of spurious and missed heart tones. A real-time working system was fabricated consisting of a sensor belt, front end electronics, a TMS320C25 digital signal processing board, an 80386 PC,

and a strip chart recorder. The apparatus allows performance of the fetal nonstress test (NST) in a manner similar to that conventionally accomplished via ultrasound. The acoustical system was implemented in parallel with a commercial ultrasound unit on a series of patients undergoing NSTs. The heart rate records are compared.

1:35

5EA3. The acoustic material signature for a cracked surface using a point focus acoustic microscope. D. A. Rebinsky and J. G. Harris (Dept. of Theoretical Appl. Mech., UIUC, 216 Talbot Lab., 104 S. Wright St., Urbana, IL 61801)

Using an asymptotic model of a point focus scanning acoustic microscope, the acoustic material signature for a cracked surface is calculated. The scattered wave fields in the coupling fluid are related to the output voltage by an electromechanical reciprocity relation. This relation may be calculated using either asymptotically approximated wave fields or their asymptotically approximated spectra. Because this relation is linear, the component of the acoustic signature produced by the crack may be separated from that produced by a defect-free surface. The component arising from the scattering from the crack is calculated from approximations to the wave fields rather than their spectra. It is assumed that the crack opening does not perturb the geometric wave field, but that it does strongly scatter the leaky Rayleigh wave, excited by the microscope. It is also assumed that scattering of the Rayleigh waves from the crack can be characterized by reflection and transmission coefficients. The wave fields scattered from the crack are thus calculated geometrically and the corresponding contribution to the acoustic signature is estimated. [Work supported by the MRC at UIUC.]

1:50

5EA4. Structural acoustic analysis using the fast Fourier transform. Peter K. Kasper (CASDE Corp., Suite 600, 2800 Shirlington Rd., Arlington, VA 22206)

Fourier analysis has long been an important tool in acoustical and structural dynamics technologies. Many of the classical problems in acoustics and dynamics can, for example, be solved in closed form using Fourier transforms. The fast Fourier transform (FFT), itself, is an efficient method for calculating the Fourier transform of discretized or sampled variables. The current paper describes a general approach using the FFT for obtaining numerical solutions of fundamental acoustics and structural dynamics differential equations. Specific examples treated include harmonic response of finite strings and beams. The general ap-

proach involves the application of a discrete Fourier transform to a Helmholtz or biharmonic equation with generalized harmonic load. For consideration of finite structures, the finite Fourier transform is used. The use of a finite Fourier transform provides a convenient way of incorporating boundary conditions. The paper presents example calculations showing the numerical solutions as compared with closed-form classical solutions. Numerical accuracy and extension to multidimensional structures are discussed.

2:05

5EA5. New approach to designing low peak factor, discrete-time waveforms. John C. Burgess^{a1} (Acoust. and Signal Processing, Inst. for Microstructural Sci., Natl. Res. Council, Ottawa, Ontario K1A 0R6, Canada)

Periodic waveforms can be designed to have the smallest possible peak (or crest) factors while having their power concentrated in a desired frequency band. A well-known approach is first to specify the magnitudes of the desired waveform harmonics, then to specify or search for a set of phase angles [M. R. Schroeder, *Number Theory in Science and Communication* (Springer-Verlag, New York, 1984); J. Pumplin, "Low-noise noise," *J. Acoust. Soc. Am.* **78**, 100-104 (1985)]. This approach requires a significant amount of computer time to search for sets of phases that provide waveforms with near-minimum peak factors. Another approach is to specify the waveform as a bandlimited, periodic chirp, $x(n) = \sin[a(n + n_0)]^2$, $0 \leq n < N$. This approach has the advantage that the waveform has the smallest peak factor possible while allowing some control of the waveform spectrum. It has the disadvantage that the spectrum is distorted by leakage. The design approach, results, and some recent efforts to reduce leakage will be discussed. [Work supported in part by National Research Council of Canada and Lanikai Foundation.] ^{a1}On leave from University of Hawaii.

2:20

5EA6. On the normal modes of free vibration of inhomogeneous and anisotropic elastic objects. William M. Visscher and Albert Migliori (Los Alamos Natl. Lab., Los Alamos, NM 87545)

The Hamilton's principle approach to the calculation of vibrational modes of elastic objects with free boundaries is exploited to compute the resonant frequencies of a variety of anisotropic elastic objects, including spheres, spheroids, ellipsoids, hemispheres, cylinders, eggs, shells, bells, sandwiches, cones, pyramids, prisms, tetrahedrons, octahedrons, and potatoes. The paramount feature of the present calculation, which distinguishes it from previous ones, is the choice of products of powers of the Cartesian coordinates as a basis set for expansion of the displacement, enabling us to analytically evaluate the required matrix elements for these systems. This algorithm will allow a general anisotropic elastic tensor with any position dependence and any shape with any density

variation. A number of plots of resonance spectra of families of elastic objects as functions of relevant parameters will be shown, and the measured resonant frequencies of a precision machined but irregularly shaped sample of aluminum (called a potato) will be compared with its computed normal modes. Possible applications to problems in materials science and in seismology will be mentioned. [Work supported by USDOE.]

2:35

5EA7. Effect of curvature discontinuities on the vibrational modes of cylindrical shells. Michael J. Bresnahan, George J. Jelatis (St. Olaf College, Northfield, MN 55057), and Thomas D. Rossing (Northern Illinois Univ., DeKalb, IL 60115)

A study has been made of the modes of vibration of three brass cylinders with almond-shaped cross sections, formed by soldering together two sections of a circular pipe. These modes appear as doublets, with the higher frequency member of the pair having a node at each discontinuity. In uncapped shells, the frequency ratio of the doublet pair decreases rapidly with increasing m (the number of bending waves around the circumference) and less rapidly with increasing n (the number of nodal circles). Frequency splittings as large as 30% were observed in the $m = 2$, $n = 0$ mode, indicating a substantial decrease in bending wave velocity at the locations where the shell segments are joined. Measurements on capped shells are in progress. [Work supported by the Pew Foundation.]

2:50

5EA8. Vibration analysis of shell structures based on a plate analogy. Jeung T. Kim (Acoust. and Vib. Lab., Korea Standards Res. Inst., P.O. Box 3, Daedok Science Town, Daejeon, Korea)

An approximated formula representing the vibration characteristics of a shell structure with finite curvature is proposed. The model is based on the dynamic analogy between a cylinder and a flat plate. In order to derive the formula, the dynamics of a shell with and without the curvature effect is introduced. Then, resonance frequencies, constant frequency loci, and number of modes in a flat plate and in a cylinder are also examined. After that, a simplified driving point mobility function of a shell with the plate mobility is derived. An experimental verification is conducted in order to examine the proposed formula. Cylinders having two different ring frequencies of 5.9 and 13.8 kHz are used as test structures. Comparisons are done for the resonance frequency and for the mobility function. The accuracy of the analysis falls within 6% error when resonance frequencies are compared with those of experimental data. The results show that the proposed mobility function predicts the vibration behavior for the frequency bandwidth below and above the ring frequency. The model is simple to use and could be easily extended to various boundary conditions for vibration and noise prediction purposes.

Session 5ED**Education in Acoustics: Demonstrations in Acoustics****Anthony A. Atchley, Cochair***Physics Department, Naval Postgraduate School, Monterey, California 93943***Ronald A. Roy, Cochair***National Center for Physical Acoustics, Coliseum Drive, University, Mississippi 38677***Chair's Introduction—1:30*****Invited Papers*****1:35**

5ED1. Psychoacoustic demonstrations of adaptation in hearing. Craig Champlin, Linda Thibodeau (Dept. of Speech Commun., Univ. of Texas, Austin, TX 78712), and Dennis McFadden (Univ. of Texas, Austin, TX 78712)

Adaptation in sensory systems involves a decline in the responsiveness to a steady stimulus with time. Three acoustic demonstrations will be presented to illustrate psychoacoustic phenomena that are thought to be related to adaptation processes. (1) Overshoot: The detectability of a brief tonal signal is often improved when its onset is delayed relative to the onset of a wide-band masker. (2) Enhancement for nonspeech sounds: The detectability of a pure-tone component in a complex sound is often improved when it is preceded by a complex in which the energy in the corresponding frequency region has been reduced. (3) Enhancement for speech sounds: Vowel-like sounds are often perceived from spectrally flat harmonic complexes when they are preceded by complexes in which the energy in the frequency regions corresponding to that vowel's formants has been reduced.

2:00

5ED2. Demonstrations for education in acoustics from laboratory research and development. Thomas B. Gabrielson (Naval Air Dev. Ctr., Code 5044, Warminster, PA 18974)

In an effort to help science teachers at local secondary schools and colleges, several classroom demonstrations have been developed as spin-offs from basic research and systems development work at the Naval Air Development Center. While this transition program is informal, the demonstrations have been well received as long as they are made simple enough for the intended class. The acoustics-oriented demonstrations include an outdoor sound-speed experiment suitable for junior or senior high school, several implosive and explosive sources of sound with a little thermodynamics, and a computer animation of the thermoacoustic cycle.

2:25

5ED3. Musical acoustics: A vehicle to enhance science teaching. Uwe J. Hansen (Dept. of Phys., Indiana State Univ., Terre Haute, IN 47809)

Workshops following the Baltimore meeting, supported with ASA "Initiative" funds, sponsored by the Technical Committee on Musical Acoustics and the Education Committee, include one section that is a direct outgrowth of a series of workshops funded with Title II funds at Indiana State University during the past several years. Workshops have been conducted at both the elementary and secondary levels with elementary classroom and music teachers participating as well as junior high school science, high school general science, physical science, and physics teachers. The object of the workshops is to use musical acoustics as a vehicle to generate enthusiasm for science. At the elementary and secondary general science levels, this is accomplished by assisting teachers to build simple instrumentation illustrating various musical resonance phenomena. Additional sophistication is added to the high school physics program by using a keyboard as an introductory Fourier synthesizer. In this presentation, the workshop program will be discussed, a simple device illustrating string and air column resonances will be demonstrated, as will be a keyboard-computer combination to show simple Fourier analysis and synthesis.

5ED4. Hydrodynamic demonstration of the classical cochlea. Robert Keolian (Dept. of Phys., Naval Postgraduate School, Code PH/Kn, Monterey, CA 93943)

The inner ear is a remarkable system for detecting and analyzing sound. To first approximation, it is a Fourier transform machine that utilizes a highly dispersive waveguide to localize the energy of each frequency of a traveling wave to a particular position. This will be demonstrated with the aid of a hydrodynamic analog. It consists of a 35-in.-long, $\frac{5}{8}$ -in.-i.d. horizontal tube (representing the scala) closed at one end and driven by a piston at the other. Projecting above this are about 100 closely spaced transparent vertical $\frac{1}{4}$ -in.-i.d. tubes (representing the cochlear partition) whose length varies exponentially from 8.25 in. near the piston to 33 in. near the closed end. Water poured into the apparatus fills the large horizontal tube, and rises up the vertical tubes to a height of 8 in. The tops of the vertical tubes are then sealed, and the air trapped above the water surface acts like a place-dependent spring. By driving the piston at various frequencies, one can see a wave in the free surface travel from the piston end toward the tube that is resonantly excited, where most of the energy will be deposited.

3:15

5ED5. Demonstrations on the scattering of sound by bubble clouds. Murray S. Korman (Dept. of Phys., U. S. Naval Acad., Annapolis, MD 20402) and Ronald A. Roy (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, University, MS 38677)

A bubblemaker (which operates by regulating a transient volume of air stored in a chamber placed between two solenoid valves) is the heart of an experimental apparatus used to generate a transient bubble cloud in water. The apparatus can be adjusted to create a cylindrical bubble column terminating at the air-water interface. Sound pulses are generated by a circular plane array "transmitting" transducer driven by an electronic tone burst generator. Backscattering, at the angle $\theta = 180^\circ$, from the cloud is measured by a receiving transducer. Results are displayed to show any difference measured between scattering by the transient cloud and by the cylindrical column. [Work supported by the Naval Academy Research Council and by the National Center for Physical Acoustics.]

Contributed Papers

3:40

5ED6. Acoustic levitation positioning of objects in water. Gunter S. Schemmann and E. Carr Everbach (Dept. of Eng., Swarthmore College, Swarthmore, PA 19081-1397)

Applications for the precise three-dimensional positioning of neutrally buoyant objects include the control of objects in space stations and the containerless processing of materials on Earth. There will be a demonstration of a microcomputer-controlled system for acoustically levitating an object and moving it to any point within a three-dimensional volume. The system uses a standing wave field in the vertical direction for levitation and a three-transducer traveling wave field to displace the object in the horizontal directions via radiation force. Pulse-echo transducers are used to provide continuous monitoring of position and a Macintosh IIx computer is used to control the position and provide user interfacing.

3:55

5ED7. Educational experiments in acoustics, signal processing, and computer data acquisition and calibration. David C. Swanson (Penn State Univ., University Park, PA 16802)

The Pennsylvania State University Graduate Program in Acoustics has recently added a Laboratory Course to its curriculum that is required for all full-time degree candidates. The course goals are to teach objective reporting skills, proper calibration of measured data, and the general use of computers and digital recordings for data analysis. Experiments included measurements of propagation changes due to temperature inversions, vibrations, reverberation time measurements, duct acoustics and impedance, source coupling, and directivity. While basic PC software was developed for operation of the A/D boards and signal processing, the experiments involving digital recording and analysis could be reproduced with a variety of commercially available software packages and computer equipment. Traceability is emphasized for all data calibration techniques so that students will understand how to verify good data records that may have been recorded by others. A summary of the particular experiments and results is presented.

Session 5MU

Musical Acoustics: Bowed Strings: Honoring Carleen Hutchins, Part 2

Uwe J. Hansen, Cochair

Department of Physics, Indiana State University, Terre Haute, Indiana 47809

Edith L. R. Corliss, Cochair

*Forest Hills Laboratory, 955 Albemarle Street, N.W., Washington, DC 20008**Invited Papers*

1:30

5MU1. Acoustics of the violin as a function of its parts. Erik V. Jansson (Dept. of Speech Commun. and Music Acoust., Royal Inst. of Technol. (KTH), P.O. Box 700 14, S-100 44 Stockholm, Sweden)

During the last 20 years, Carleen Hutchins has successfully developed methods for testing and tuning of free top and back plates with Chladni patterns. Her methods have been successfully used by other makers in Sweden too. For the engineer/physicist, it is, however, difficult to understand how and how much of the quality of a violin is predicted by the free-plate tuning. Therefore, data from KTH experiments have been used to investigate relations between free plates and assembled violins. Eigenmodes of free top and back plates have been calculated and measured. Effects of thinning the top plate free and in an assembled violin have also been investigated. Comparison of the frequency of the main resonance of the violin (T1 at about 500 Hz with mainly the top plate vibrating along the bass bar side) is affected somewhat similarly to thinnings of the ring mode of the free plate (i.e., it is most sensitive to off-center thinnings and it should be a balance in stiffness along and across). Thus the present result is qualitatively in agreement with that of Dr. Hutchins—the ring mode of the free plate is the most important for tuning. Recent playing tests imply, however, that the lengthwise stiffness of the top plate is more important than the one across. [Work together with Luleå Technical University, L. Frydén, Stockholm, J. and B. Niewczyk, Poznan.]

2:00

5MU2. The influence of the bow on aperiodicity of violin notes. Robert T. Schumacher (Dept. of Physics, Carnegie Mellon Univ., Pittsburgh, PA 15213)

A method has been demonstrated to investigate the aperiodicity in nearly periodic signals [R. T. Schumacher and C. Chafe, ICASSP-90, Paper 6.A1.18, Albuquerque, NM (1990)]. This method, the norm difference method, quantifies the fluctuations in the shapes of waveforms from cycle to cycle. The norm difference method will be explained. As an application, a study of the difference in aperiodicity of a note separately played on a violin by two bows will be presented. In addition, a brief survey of the range of aperiodicities exhibited by other orchestral instruments will be presented in order to compare them with characteristic bowed string aperiodicities.

2:30

5MU3. The dynamics of musical strings. Maurice Hancock (66 Salisbury Rd., Farnborough, Hants. GU14 7AG, England)

The resonance contours of a number of musical strings have been traced when operating in isolation from an instrument body, with more precisely defined loading and terminal conditions than are found in normal use, and with very small excitation amplitudes. For plain steel piano strings, results from the fundamental to the 25th overtone are broadly in accordance with the predictions of the standard classical theory for a lossy stiff string, but close agreement with calculated data cannot be obtained for any one set of the relevant parameters, and some disturbing influences are clearly in operation. Some of the resonances show double peaks suggestive of close coupling between slightly dissimilar modes, and for these and the single peaks, associated transverse resonances perpendicular to the line of the driving force are often found. A copper wire string hammered to produce an exaggerated linear distribution of deformation shows a more pronounced pattern of double and transverse resonances, and it is surmised that these occurrences with the steel strings are due to small departures from exact cylindrical uniformity. Overwound strings show similar anomalies with significantly higher losses.

5MU4. Acoustical significance of the violin octet. Carleen M. Hutchins (Catgut Acoust. Soc., Inc., 112 Essex Ave., Montclair, NJ 07042)

The 30-year acoustical and musical development of the violin octet has shown the following. (1) New instruments of the violin family can be created with fine tone and playing qualities based on acoustical parameters, free-plate tuning, and skilled violin making. (2) The prime controlling factor differentiating overall tone quality of the violin from that of viola, cello, and string bass is frequency placement in relation to string tuning of the body length air cavity mode (A_1) originally called the "main wood" resonance. (3) The secondary controlling factor for tone quality, especially on the two lower strings is frequency placement in relation to string tuning of the A_0 cavity mode originally labeled "Helmholtz" or "main air" resonance. (4) For instruments smaller than the cello, the A_0 cavity mode frequency is controlled more by air volume than by compliance of the top, back, and ribs (sides). (5) For cello and larger instruments the A_0 cavity mode frequency is controlled more by compliance of the top, back, and ribs than by air volume. (6) Due to greatly increased compliance of the wooden walls of basses, it has been found structurally unsafe to make ribs shallow enough to place the A_0 cavity mode seven semitones above the lowest note as our scaling theory projected originally. These findings will be discussed and illustrated.

Contributed Papers

3:30

5MU5. Dynamic mechanical properties of violin wood. Edwin R. Fitzgerald (102 Latrobe Hall, Johns Hopkins Univ., 34th and Charles Sts., Baltimore, MD 21218)

Values of complex shear compliance ($J^* = J' - iJ''$) and modulus ($G^* = 1/J^*$) have been measured for spruce and maple over a continuous frequency range from 2–10 000 Hz, and at temperatures from 15 to 40 °C. The wood strip samples of European spruce and Norway maple were supplied by Carleen M. Hutchins in connection with her investigations of the effect of various acoustical parameters of wood on plate tuning and the tone qualities of finished violins. Measurements were made in an automated electromagnetic transducer apparatus [E. R. Fitzgerald and R. E. Fitzgerald, *Polymer Bull.* **18**, 167–174 (1987)] in which samples are sheared while clamped between stainless steel blocks. Values of the shear parameters vary with frequency, but also with the grain orientation, moisture content, and the perpendicular, clamping force on the sample faces while they are vibrated in shear. The mechanical spectra differ, but, in general, several sharp, microstructural compliance modes are superimposed on broad retardation, background spectra [E. R. Fitzgerald, *J. Acoust. Soc. Am.* **33**, 1305–1314 (1961)]. Typical are the results for a spruce sample at 22 °C, sheared cross grain, for which values of elastic compliance (J') decrease from 2.74 to 0.306×10^{-9} cm²/dyn (2.74 to 0.306×10^{-2} MPa⁻¹) as the frequency increases from 10 to 10 000 Hz; the loss compliance (J'') rises to a broad maximum of 1.65×10^{-9} cm²/dyn (1.65×10^{-2} MPa⁻¹) at

4000 Hz. Values of the shear sound velocity and attenuation, together with the mechanical loss tangent ($J''/J' = G''/G'$), are also calculated for the samples.

3:45

5MU6. Quasiperiodicity and bifurcations in wolf tones. René Caussé (IRCAM, 31 rue St-Merri, 75004 Paris, France), Jean Puaud (Univ. du Maine, 72017 Le Mans Cedex, France), and Vincent Gibiat (ESPCI/Université Paris 7, 75231 Paris Cedex 05, France)

The wolf tone, often obtained on the lower bowed stringed instrument, is studied with an experimental setup that mimics the true instrument with the help of a digital bow [Caussé and Weinreich, *Proceed. 13th ICA*, Belgrade 1989]. In this experiment, the resonant frequency of the bridge can be adjusted so that it is low enough for a good coupling with the fundamental frequency of the string, while keeping the bridge as rigid as possible. With this experimental setup, verification of the well-known result that the wolf tone depends strongly on the pressure and speed of the bow and on the bowing point of the string has been made. By changing bow speed and pressure, by bifurcations after the normal periodic sound (Helmholtz motion), various quasiperiodic tones built on two or three basic frequencies have easily been obtained. Other possible scenarios related to bifurcation theory are indicated by observations of more complex signals.

Session 5PA

Physical Acoustics: Nonlinear Phenomena

Andres Larraza, Chair

Physics Department, Naval Postgraduate School, Monterey, California 93943

Contributed Papers

1:30

5PA1. The trans-spectral coherence in the solution of the Duffing's equation, undergoing chaos. P. G. Vaidya (Dept. of Mech. and Materials Eng., Washington State Univ., Pullman, WA 99164)

An analysis of the time series generated by the Duffing's oscillator, undergoing chaos, has been carried out. The series is inspected for the existence of the trans-spectral coherence, which is the coherence between two different spectral components. (The technique to calculate such coherence has been demonstrated previously by Vaidya and Anderson.) In the case of the Duffing's equation, a significant coherence is seen across the trans-spectrum. This shows the presence of order, co-existing with randomness, in this series. A companion paper at this conference explains the origin of such order.

1:45

5PA2. Existence of a hierarchy of periodic components in the solution of the Duffing's equation, undergoing chaos. P. G. Vaidya and Amit Athalye (Dept. of Mech. and Materials Eng., Washington State Univ., Pullman, WA 99164)

Chaos is recognized as a state manifesting the characteristics of both order and randomness. An analysis of the Duffing's oscillator, undergoing chaos, has been carried out. The solution is written as a sum of a periodic solution, representing a limit cycle, and a residue. When the residue is assumed to be small, it is shown to be governed by a linear nonautonomous equation that is unstable. When the residue is allowed to be large, it is governed by an equation whose solution is once again bounded and chaotic. This new solution can, in turn, be represented as the sum of a periodic and a chaotic solution. This process in theory can be carried out indefinitely. The analysis sheds some light on the nature of chaos generated by the Duffing's equation, and it also explains the significant amount of the trans-spectral coherence observed in the time series generated by its solution.

2:00

5PA3. A version of NPE for nonlinear propagation of ultrasonic pulses in focused field. Gee-Pinn James Too, Jerry H. Ginsberg, and Jacqueline Naze Tjøtta (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332)

The NPE (nonlinear progressive wave equation) and associated computer program are a time domain representation that was developed by McDonald and Kuperman to study waveguide problems. The present work extends the earlier studies [Too and Ginsberg (1989-1990)], which developed a modified version of NPE in terms of axisymmetric cylindrical coordinates to describe propagation of finite-amplitude sound beams. In the present studies, a new version of NPE is studied in which axisymmetric spherical coordinates are employed to

describe propagation of axisymmetric convergent and divergent waves. The present problem concerns nonlinear pulse propagation in a focused field. In order to initialize the moving window convected by NPE, a linear spherical wave assumption is made adjacent to the transducer; i.e., inside the converging beam, the input field is represented as a linear spherical wave, while outside the beam, the input field is considered to be zero. Temporal waveforms are computed on- and off-axis and are compared to the experimental data by Baker and Humphrey [Frontiers of Nonlinear Acoustics 12th ISNA, pp. 185-190 (1990)]. [Work supported by George W. Woodruff endowment.]

2:15

5PA4. Harmonic generation in focused sound reflected from a curved surface. Inder Raj S. Makin (Biomed. Eng. Prog., The Univ. of Texas at Austin, Austin, TX 78712-1084) and Mark F. Hamilton (The Univ. of Texas at Austin, Austin, TX 78712-1063)

In an earlier presentation [A. Averkiou and M. F. Hamilton, J. Acoust. Soc. Am. Suppl. 1 **85**, S93 (1989)], an analysis of the linear propagation and reflection of a focused Gaussian beam was reported. The reflecting surface was assumed to be slightly curved and perfectly rigid. Here, second harmonic generation in the incident and reflected beams is investigated, with absorption and finite surface impedance now taken into account. Closed-form solutions for the second harmonic pressure and power are derived from the KZK nonlinear parabolic wave equation. Propagation curves and beam patterns are presented for various target curvatures and impedances. Both before and after reflection, the transverse distribution of the second harmonic pressure is equal to the square of the transverse distribution of the fundamental pressure. Variations in the relative phase between the fundamental and second harmonic components, due to propagation through foci and reflection from the target, significantly influence the process of harmonic generation. Implications for ultrasonic imaging are discussed. [Work supported by NSF, the David and Lucile Packard Foundation, and the Texas Advanced Research Program.]

2:30

5PA5. Finite-amplitude wave propagation through a two-phase system using coupled generalized Burgers' equations. A. Benharbit (Dept. of Math., Penn State Univ.—York Campus), T. S. Margulies (U.S. Nuclear Regulatory Commission, Washington, DC 20555), and W. H. Schwarz (Johns Hopkins Univ., Baltimore, MD 21218)

The propagation of finite-amplitude acoustic waves through a system of fluid particles in a fluid matrix (aerosols or emulsions) has been examined theoretically by using the continuum volume-averaged balance equations and linear constitutive equations for a two-phase system. Utilizing the technique developed by Lighthill for clean perfect gases, and by Davidson [G. A. Davidson, J. Sound Vib. **38**, 475-495 (1975)],

who employed the equations of Marble to dilute aerosol media, coupled Burgers' equations for each phase were obtained and analyzed by a perturbation method. The behavior of the higher harmonics of an initial sinusoidal plane wave was determined, and compared to previous work. The present results are applicable to moderately concentrated particulate systems. Also, the use of two-phase equations obtains the effects of viscosity of the media, and also phoresis, such as thermophoresis, on the waveforms.

2:45

SPA6. Nonlinear wave propagation in reacting, viscoelastic fluids. T. Margulies, J. Randall (U.S. Nuclear Regulatory Commission, Office of Res., Washington, DC 20555), and W. H. Schwarz (Johns Hopkins Univ., Baltimore, MD 21218)

Finite-amplitude wave propagation in a viscoelastic fluid has been investigated with a perturbation approach using multiple time scales. A generalized Burgers' equation (GBE) for planar and nonplanar (i.e., cylindrical and spherical) waves has been developed for a continuum, simple fluid representation of Coleman and Noll. Simultaneous chemical reactions are coupled in the analysis by modifying the pressure perturbations for a reacting system with orthonormalized reaction progress variables. The dissipation in the analysis results from both stress relaxation and chemical relaxation, as well as heat transfer by Fourier conduction. The GBE with an integral stress relaxation functional reduces to the classical Burgers' equation form for the case of a Newtonian viscous, perfect gas without reaction. Other approximations and limitations are discussed, including, for example, approximations of nonconvex equations of state, and derivations of a Korteweg de Vries-Burgers' equation exhibiting both dissipation and dispersion.

3:00

SPA7. Linear and nonlinear propagation of a pulsed sound beam in a dissipative fluid. Kjell-Eivind Frøysa (Dept. of Math., The Univ. of Bergen, Allegt. 55, 5007 Bergen, Norway)

Linear and nonlinear propagation of pulsed sound beams is considered. The combined effects of diffraction, absorption, and nonlinearity are discussed within the linear and the weakly nonlinear (quasilinear) approximation of the Khokhlov-Zabolotskaya-Kuznetsov nonlinear parabolic equation. For the special case of a Gaussian on-source amplitude distribution, short pulses can be followed all the way from the sound source and into the dissipative far field, both within a linear and a weakly nonlinear model. The results obtained are related to the theory of scattering of sound by sound [Berntsen *et al.*, J. Acoust. Soc. Am. **86**, 1968-1983 (1989)], and to the self-demodulation of transient carriers [H. O. Berklay, J. Sound Vib. **2**, 435-461 (1965)]. [Work supported by the Norwegian Research Council for Science and Humanities.]

3:15

SPA8. Comparison of numerical and analytical treatments of the transient parametric array. William Hogan, Harvey Woodsum (Sonotech Corp., 47 Constitution Dr., Bedford, NH 03102), and Peter J. Westervelt (Sonotech Corp. and Brown Univ., Providence, RI 02912)

Recently, Westervelt [J. Acoust. Soc. Am. Suppl. **1** **88**, S167 (1990)] has developed an analytical treatment of the transient parametric array that is valid for both far field and near field, for arbitrary taper function and arbitrary modulation of the primary wave. The result was previously shown to be consistent in the far field with results of P. Stepanishen [J. Acoust. Soc. Am. **82** (1987)]. The analysis is based on earlier work in general relativity by Westervelt [Acta Phys. Pol. **27**, 831-841 (1965)]. Here, a useful numerical treatment of the same problem is presented. A computer model for this numerical formulation has been implemented, and results have been obtained to compare with Westervelt's analytical results. Comparison is also made between numerical results and relevant experimental data found in the literature.

WEDNESDAY AFTERNOON, 1 MAY 1991

INTERNATIONAL C, 1:00 TO 4:05 P.M.

Session 5PP

Psychological and Physiological Acoustics: Session Honoring J. Donald Harris

A. Yonovitz, Chair

Conley Speech and Hearing Center, The University of Maine, Orono, Maine 04469

Invited Papers

1:00

5PP1. J. Donald Harris: The humanist and the scientist. A. Yonovitz (Conley Speech and Hearing Ctr., Univ. of Maine, Orono, ME 04469)

J. Donald Harris was born in Brownsburg, Pennsylvania. From an early age, his thoughts and efforts have been devoted to the auditory world and its significance to the human condition. In 1942, he received his Ph.D. in psychology from the University of Rochester. He began conducting research and became head of the Auditory Research Laboratory at the Naval Submarine Medical Research Laboratory in Groton, Connecticut. He continued in that position until his retirement in 1979. He maintained appointments with the Doctoral Program at the City University of New York and the Departments of Speech and Psychology at the University of Connecticut. During his career, he published over 160 articles, including chapters,

books, and monographs on subjects ranging from physiological and psychological acoustics to audiology and education. He was the Editor in Chief of the *Journal of the American Auditory Society (Ear and Hearing)*. J. Donald Harris was also the founder and editor of the *Journal of Auditory Research*. It provided a vehicle for dissemination with a philosophy embracing interdisciplinary acoustics and the advancement of the sciences of hearing. Don provided a modiolus for the student, researcher, and clinician to relate the needs of the human to the research laboratories in hearing. His writing is a gift to those whose efforts are designed to infuse humanitarian sentiments in science. He will be greatly missed.

1:20

SPP2. My love affair with Ruth Bender. Barbara Franklin (Dept. of Special Education, San Francisco State Univ., 1600 Holloway Ave., San Francisco, CA 94132)

In 1982, *The American Auditory Society* saluted J. Donald Harris as its Renaissance Man. His unique ability to combine science and the humanities is exemplified in his chapter, "My love affair with Ruth Bender: A history of binaural aids for babies" [Harris, in *Binaural Hearing and Amplification*, edited by Libby (Zenetron Inc., Chicago, 1980)]. Binaural hearing was a subject very dear to his heart and he praised the pioneering work of Ruth Bender and others in fitting infants and young children with two hearing aids. Dr. Harris served as mentor and major advisor to the writer during the time he was an adjunct professor at CUNY. He supervised her dissertation that compared discrimination scores when a low- and high-pass band of speech was presented to the same ear or opposite ears of normal-hearing subjects. This study was subsequently replicated by the writer with hearing-impaired individuals, both congenital and adventitious. This paper will present an overview of this research and will conclude with a study that investigated the use of split-band amplification—high frequencies in one ear and low frequencies in the other. The writer received ongoing advice, encouragement, and support from J. Donald Harris for her research. The passing of our Renaissance Man has left an unfillable void for many.

1:40

SPP3. Acoustic processing as a basis for navigation by divers. Harry Hollien (Inst. for Advanced Study of the Commun. Processes, Univ. of Florida, Gainesville, FL 32611)

Divers traditionally experience substantial difficulty when they attempt to navigate distances underwater. In air, of course, they enjoy good vision plus other sensory cues to accomplish this task. However, when submerged, the diver's visual modality is sharply impaired and, in a sense, he or she is left virtually blind. In many cases, divers attempt to navigate by means of compass settings (dead reckoning), but relevant research has demonstrated that this approach leads to unacceptable errors. Thus it appeared that some other, more efficient, approach needed to be developed. A number of experiments on this topic have been conducted and reported; the overall program was stimulated to a great extent by interface with Donald Harris. The basic approach employed focused on the abilities of divers to navigate by responding to programmed acoustic signals. That is, it had been noted that sound can be made to (perceptually) "move" underwater if it is serially shifted from transducer to transducer. This perception is referred to as the underwater auditory phi phenomenon (UAPP); it greatly aids in sound localization and, ultimately, navigation. Indeed, for diver retrieval, this phenomenon is so powerful that no subject in any of these experiments has ever swum to an area except that from which the source signal emanated. Previously published data will be reviewed briefly; new data on the effects of training and error magnitude will be presented and discussed.

2:00

SPP4. Experiments in binaural hearing: Masking, speech intelligibility, and binaural hearing aids. Harry Levitt (Ctr. for Res. in Speech and Hearing Sci., City Univ. of New York Graduate School, 33 West 42nd St., New York, NY 10036)

The late J. Donald Harris had a great interest in binaural hearing and binaural amplification systems for the hearing impaired. This paper is dedicated to his memory. Research on binaural release from masking has been substantial. A similar but smaller research effort has focused on binaural improvements in intelligibility and related issues involving binaural hearing aids. There have been few attempts, however, to integrate the two areas of research. Methods for predicting binaural improvements in intelligibility from models of binaural release from masking will be reviewed. Data from an experimental evaluation of one such method will be presented and the implications for the design of binaural amplification systems for the hearing impaired will be discussed.

SPP5. The importance of overload distortion in hearing aids. Mead C. Killion (Etymotic Res., 61 Martin Ln., Elk Grove Village, IL 60007)

In a series of three separate experiments using a total of 5 talkers, 3 speech conditions, 20 normal and 20 hearing-impaired subjects, and 30 different hearing aid–receiver combinations, Harris and his colleagues concluded; “All three experiments show harmonic distortion, alone of all the several electroacoustic characteristics studied, to affect intelligibility in any really significant amount [J. Aud. Res. 1, 357–381 (1961)].” This paper will summarize (a) recent circuit developments that have made possible a reduction in both input-circuit and output-circuit overload distortion to inaudible levels (even for full-orchestra fortissimo inputs), and (b) recent indications that the distortion-reduction capabilities of automatic signal processing (ASP) circuits may be more important than their level-dependent-frequency-response-shaping capabilities. Indeed, ASP circuits that do opposite things (decreased high-frequency or decreased low-frequency gain for high-level inputs) are both successful in the marketplace.

2:40–3:00

Break

3:00

SPP6. The nature of the generator of outer hair cell motility. Burt N. Evans, Richard Hallworth, and Peter Dallos (Auditory Physiol. Lab., The Hugh Knowles Ctr., Northwestern Univ., 2299 Sheridan Rd., Evanston, IL 60208)

Despite the lack of obvious substrates underlying the electrokinetic shape changes of mammalian outer hair cells (OHCs), much can be learned about the mechanism when cells are electrically and mechanically partitioned in a glass microchamber. The results of such studies employing photometric as well as video analysis support the notion that the cellular motor consists of a large number of small, independent elements restricted to the supranuclear region of the cell's cortex. The length changes measured in OHCs are highly nonlinear. They are subject to manipulation as well as vulnerable to overstimulation. Thus the mechanical response resembles the mechanism underlying sharp tuning in the intact cochlea. Interestingly, such nonlinearities can be shown to be inherent to each voltage-sensing element. The whole-cell movement is thus effectively the cumulative response of many such elements, each driven in parallel by the local transmembrane potential along the length of the cell.

Contributed Papers

3:20

SPP7. Speech intelligibility for normal and hearing-impaired individuals under “everyday” conditions of acoustic distortion. Paul G. Lacroix (Navy Environmental Health Ctr., Detachment C/O Naval Medical Clinic, Portsmouth, NH 03804-5000)

The intelligibility of speech in daily life is often made difficult when acoustic distortions are either produced at the source by the talker, or induced elsewhere in the process of transmission to the listener. Many variables must be defined, ordered, and controlled before the relative and combined contribution of acoustic distortions under “everyday” conditions can be examined. The task of attacking such a multivariate problem is so daunting as to discourage all but the most ambitious researchers, but the benefit of obtaining information so relevant to the handicapping effects of hearing loss in the real world led J. D. Harris to set his formidable intellect to the task with vigor and fervor. This paper explores his views and approach to answering questions such as: What constitutes the domain of “everyday” acoustic distortions? How should speech intelligibility be measured to capture real world validity and utility? What effects are produced by “everyday” acoustic distortions when the listener is hearing impaired? Can a valid sample of “everyday” listening conditions be developed?

3:35

SPP8. Faith in natural systems. E. Robert Libby (Associated Hearing Instruments, 6796 Market St., Upper Darby, PA 19082)

The faith in natural systems principle holds that if an engineering effort is being directed at duplicating or simulating a natural system, that effort is most efficiently directed at incorporating critical functions and structures of that system. The faith in natural systems principle would say that if a person would want to design a more effective hearing instrument, that effort is most efficiently directed at incorporating the critical functions of the hearing mechanism, and the enhancement of the anatomical structures of that natural system. Binaurality is supportive of the faith in natural systems principle. The choice of an in-the-ear microphone location, ear canal resonances, and compensation for the recruitment phenomenon are also supportive of the faith in natural systems principle. The use of real ear measurements utilizing a probe tube microphone will demonstrate support of these principles.

3:50

SPP9. Circadian rhythm-dependent gentamicin-induced hearing loss in rodents. Mary Roy, Coleen Langill, and A. Yonovitz (Conley Speech and Hearing Ctr., Univ. of Maine, Orono, ME 04469)

The use of gentamicin as an antimicrobial agent has been shown to produce as an untoward effect, ototoxicity. The purpose of this study was to investigate differential effects of gentamicin ototoxicity as a function of Rx timing with regard to circadian rhythms. Sprague-Dawley rats received a daily subcutaneous dosage of 100 mg/kg of gentamicin. The rats were maintained on a light-dark 12:12 illumination cycle with light commencing at 0000. One group of rats received gentamicin at 0200 with the other group at 1400. Hearing loss was assessed with the

auditory brainstem response using pure-tone stimuli of 8, 16, 24, and 32 kHz. These measures were obtained at baseline, 2 and 4 weeks after the initial dosing. Measures of blood-gentamicin level and nephrotoxicity were contrasted for the groups. Ototoxicity was greater for gentamicin when given to the rodents during their diurnal rest span (0200) in comparison to their nocturnal activity span (1400). These circadian dependent effects are thought to be related to rhythmic variations in the disposition and accumulation of gentamicin in cochlear tissues.

WEDNESDAY AFTERNOON, 1 MAY 1991

INTERNATIONAL D, 1:45 TO 4:05 P.M.

Session 5SA

Structural Acoustics and Vibration: Structural Acoustics and Fluid-Structure Interaction

Jerry H. Ginsberg, Cochair

School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, Georgia 30332

David Feit, Cochair

Code 1904, David Taylor Research Center, Bethesda, Maryland 20084

Chair's Introduction—1:45

Contributed Papers

1:50

5SA1. Phase-space description of random fluid-structure interaction. E. K. Dimitriadis, J. J. McCoy, and M. J. Beran (School of Eng. and Architecture, Catholic Univ. of America, Washington, DC 20064)

Random dynamic excitation of plates in contact with fluids generally extends over a finite region of the plate. Consequently both excitation and response fields are spatially inhomogeneous. Two-point correlations of such fields are here described as functions of an average and a distance coordinate. The latter is subsequently Fourier transformed and the resulting wave-vector spectrum $\Gamma(\mathbf{x}, \mathbf{k}; \omega)$ (often called the Wigner distribution function), defined simultaneously in physical and wave-vector space, namely a phase-space, loosely describes energy flow at \mathbf{x} associated with the length scale and direction of \mathbf{k} . It is shown that the Wigner functions for the response for the excitation are related through a convolution with a kernel function $G_r(\mathbf{x} - \mathbf{r}, \mathbf{k}; \omega)$ that depends only on a small number of nondimensional system parameters. The nature of this kernel function and the resulting energy flow directivity are discussed and interpreted. In addition, results are presented for a specific example of an infinite plate in contact with a heavy fluid and excited by a finite region of turbulence in the fluid.

2:05

5SA2. Observable-based hybrid ray-mode-resonance system parametrization of acoustic scattering by a finite submerged steel plate. L. B. Felsen^{a)} and T. K. Kapoor (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

Numerical finite difference data for pulse scattering by a submerged steel plate of finite width, when processed in the slowness-time domain [J. R. Fricke and A. B. Baggeroer, J. Acoust. Soc. Am. Suppl. 1 88, S51 (1990)], reveals distinct features related to the longitudinal and flexural modes in the plate, their excitation by, and conversion to, edge-

diffracted fields in the fluid, and other wave-oriented observables. By an observable-based parametrization (OBP), the problem is phrased here systematically and phenomenologically in terms of ray fields, traveling mode fields, edge coupling matrices, resonances, etc., to build up a self-consistent OBP system format for quantitative prediction. Determination of the elements in the coupling matrices poses a set of canonical problems that can be addressed analytically under highly simplified conditions but must generally be done numerically or by experiment. For the very specialized case of a thin plate under heavy fluid loading, the plate supports only one subsonic wave, and single-edge diffraction has been evaluated analytically [D. C. Crighton and D. Innes, Philos. Trans. R. Soc. London Ser. A 312, 291-341 (1984)]. This solution has been employed in the OBP algorithm to synthesize and compute the far-field acoustic response to on-plate line force excitation of a finite strip. The result is compared with a differently synthesized solution by Crighton. [Work supported by ONR.] ^{a)}Visiting.

2:20

5SA3. Acoustic radiation from fluid-loaded structures with discontinuities. Charles Thompson, Vineet Mehta, and Rahul Sen (Dept. of Elec. Eng., Lab. for Advanced Computation, Univ. of Lowell, Lowell, MA 01854)

The presence of discontinuities, such as joints, in a fluid-loaded structure results in far-field acoustic radiation even when subsonic wave numbers prevail in the main body of the structure. The object of this study is to develop a rational model of this phenomenon using singular perturbation methods. When a flexural wave in a thin structure impinges on a joint, evanescent structural displacement fields are set up in the vicinity of the joint. It is proposed to show that these fields are associated with rotation-dominated modes that are supported by shear-corrected plate theories of the Timoshenko-Mindlin-type flexural mode. The method of matched asymptotic expansions will be used to obtain

mode-conversion coefficients, and it will be shown that the spatial wave-number spectrum of the local field is instrumental in setting up an acoustic far field in the surrounding fluid. An important mathematic issue is the proper description of evanescent fields; this analysis will be based on results obtained previously in connection with acoustic waveguides.

2:35

5SA4. Surface variational principle analysis of the response of an eccentrically loaded submerged disk in a baffle. Jerry H. Ginsberg and P. T. Chu (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332)

As an initial step in the extension of the surface variational principle (SVP) to an assumed mode analysis of vibratory displacement and surface pressure on submerged axisymmetric structures subjected to arbitrary nonsymmetric loading, the present study considers a harmonic point force applied at an arbitrary location of an elastic plate supported by an infinite, rigid baffle. Fourier series expansions of the azimuthal dependence of pressure and displacement are shown to be uncoupled, with each harmonic being governed by equations that are similar in form to those for the analogous axisymmetric problem [J. H. Ginsberg, P. T. Chen, and A. D. Pierce, *J. Acoust. Soc. Am.* **88**, 548–559 (1990)]. Recursion relations using coefficients developed in the course of solving the axisymmetric problem are shown to substantially expedite the evaluation of the additional azimuthal harmonics. Results for a force located at $r=a/2$ when $ka=3.35$, where a is the radius of the plate, are presented in terms of the radial variation associated with each harmonic, as well as overall surface distributions. It is shown that $m=0$ to $m=4$ azimuthal harmonics are comparable in magnitude, and that other harmonics are insignificant. In addition, radiated power is evaluated as a function of m . [Work supported by ONR, Code 1132-SA.]

2:50

5SA5. Waves on fluid-loaded thin elastic plates: Analytical study based on full elastodynamic equations. Allan D. Pierce and Martin G. Manley (Graduate Prog. in Acoust. and Dept. of Mech. Eng., Penn State Univ., 157 Hammond Bldg., University Park, PA 16802)

An elastic, infinite plate of finite thickness with fluid loading on one side and vacuum on the other is considered. Displacement of the plate surface is derived from the full elastodynamic equations, rather than from a plate model. Following Crighton's parametrization scheme, analytical expressions of the dispersion and polarization relations of the bending wave are derived. Systematic expansions of the dispersion and polarization relations are developed using a symbolic manipulation computer program. [Work of ADP is supported by ONR and by the William E. Leonhard endowment to Penn State Univ.; work of MGM is supported by the PSU Appl. Res. Lab. Exploratory and Foundational Res. Prog.]

3:05

5SA6. Numerical implementation of ray-acoustic algorithm for sound wave scattering by submerged elastic spherical shells. J. M. Ho, X. J. Gao, and L. B. Felsen (Elec. Eng. Dept., Weber Res. Inst., Polytech. Univ., Farmingdale, NY 11735)

The rigorously derived ray-acoustic algorithm for source-excited fluid-loaded thin elastic spherical shells [J. M. Ho and L. B. Felsen, *J. Acoust. Soc. Am.* **88**, 2389–2414 (1990)] is implemented numerically for far-field plane-wave and point-source scattering, and is compared with reference solutions based on exact spherical harmonic expansions. These comparisons permit an assessment of the range of applicability of thin-shell theory, and they indicate when and how [cf. S. P. Kargl and P. L. Marston, *J. Acoust. Soc. Am.* **85**, 1014–1028 (1989)] the algorithm should be modified phenomenologically to accommodate thick

shells. The generally good agreement between the observable-based ray-acoustic algorithm and the reference solution provides further confirmation of the utility of the physically incisive ray acoustic parametrization for this canonical problem. Some preliminary results are presented for applying the ray-acoustic scheme to a rigidly baffled hemisphere. [Work supported by ONR and DTRC.]

3:20

5SA7. Finite-element analysis of airfoil vibration using variable thickness plate elements. Paul J. Zoccola and Yun-Fan Hwang (Flow Noise Branch, David Taylor Res. Ctr., Bethesda, MD 20084)

A NASTRAN finite-element model has been developed for calculating the mode shapes and frequencies of a cantilever beam with airfoil cross section. This model uses QUAD4 elements and takes advantage of the feature allowing a different thickness at each node on the element. This feature is particularly attractive for modeling the trailing edge (wedge) section of the airfoil. Numerical results are compared with those from a model of the same beam using solid (HEXA1) elements and with an experimental modal analysis. For bending modes, the results from the QUAD4 and HEXA1 models agreed well with experimental results but the HEXA1 model required a much greater number of nodes and much more computer time. For torsional modes, however, a discrepancy exists in the calculated modal frequencies between the QUAD4 and HEXA1 models. Reasons for this discrepancy are discussed. The chordwise modal behavior of the airfoil and its influence on the vibroacoustic response are also discussed.

3:35

5SA8. Wave propagation below the ring frequency on fluid-loaded cylindrical shells. Steven L. Means (Graduate Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

Williams, Houston, and Bucaro [*J. Acoust. Soc. Am.* **87**, 513–522 (1990)] have published experimental and theoretical helical wave spectra for a point-driven, fluid-loaded shell. A previous attempt (qualitatively successful) by Kouzoupis [*J. Acoust. Soc. Am. Suppl.* **1** **87**, S163 (1990)] to explain these spectra in terms of the wave theory of structural acoustics was based on a model that neglected fluid loading in first approximation, but which included it as a correction in a somewhat simple manner. The present paper reexamines the problem with a wave theory of shells that takes fluid loading into account at the outset. The wave theory interprets the helical spectra as a plot within the wave-number plane (k_x vs k_y) for fixed angular frequency ω , derived from a dispersion relation $F(k_x, k_y, \omega) = 0$, whose form has nothing to do with the manner of excitation. The more complete analysis produces curves that have shapes that resemble a figure 8, when ω is less than the ring frequency, but a simplified analysis that takes the shell to be arbitrarily thin produces two hyperbolas. The top and the bottom of the eight are sensitive to the shell thickness, and an asymptotic theory is described that shows just how this feature varies with thickness. [Work supported by ONR and by the William E. Leonhard endowment to Penn. State Univ. The author acknowledges the advice of A. D. Pierce.]

3:50

5SA9. A structural modeling of torsional ship motions. M. Cengiz Dökmeçi (Dept. of Naval Architecture, Istanbul Technical Univ., P. K. 9, Taksim, Istanbul 80191, Turkey)

This study is concerned with a structural modeling for torsional motions of thin-walled girders of ships by beam idealization on the basis of three-dimensional theory of elastodynamics. In the modeling, (1) the fundamental equations of elastodynamics are expressed in a unified variational form that is expressed by means of Hamilton's principle through Friedrichs's transformation [M. C. Dökmeçi, *IEEE Trans. UFGC-37*, 369–385 (1990)]. Next, (2) a series representation in the aerial coordinates of cross section is introduced for the displacement field. Mind-

lin's averaging procedure in the variational form is used so as to derive the one-dimensional governing equations of torsional motions. The governing equations that are expressed both in local and variational forms incorporate all the significant effects of motion; the warping of cross section, the geometrical nonlinearity, and the fluid effect by the added

mass concept of ship are all taken into account. Further, (3) special cases are discussed and, in particular, the fully linearized governing equations are studied. The sufficient conditions are enumerated for a unique solution of the linearized governing equations by use of the logarithmic convexity argument.

WEDNESDAY AFTERNOON, 1 MAY 1991

INTERNATIONAL B, 1:30 TO 4:30 P.M.

Session 5SP

Speech Communication and Psychological and Physiological Acoustics: SP and PP Potpourri (Poster Session)

Doug H. Whalen, Chair

Haskins Laboratories, 270 Crown Street, New Haven, Connecticut 06510

All posters will be on display from 1:30 to 4:30 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30 to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 to 4:30 p.m.

5SP1. Acoustic target zones for naturally produced vowels in running speech. Frank E. Kramer, Donald J. Meyer, Marios Fourakis, and James D. Miller (Central Inst. for the Deaf, 818 S. Euclid Ave., St. Louis, MO 63110)

Acoustic measures of 2304 vowel tokens of Midwestern American English were collected by one of us (Fourakis). The vowels [i, ɪ, e, æ, ʌ, ɔ, ʊ, u, ʌ] occurred in [bVd] or [hVd] syllables imbedded in sentences spoken with different rates of speech and different stress patterns. There were four male and four female speakers. Two methods of automatic classification of this corpus of acoustic measures into vowel categories are evaluated. One method derives from parametric statistical theory and has been used often in studies of speech sounds. This method is linear discriminant analysis (LDA). The other method assigns each point in the acoustic space a most likely "vowel classification" by a "spatial windowing" procedure. Then an edge-detection algorithm automatically finds the edges of regions with similarly classified interior points. This is called the target-zone method (TZM). Both methods are applied completely automatically to the acoustic descriptions of the vowels. When the vowels are described acoustically in terms of Miller's auditory-perceptual space [J. D. Miller, *J. Acoust. Soc. Am.* **85**, 2114–2134 (1989)], the target zones automatically generated by TZM classify about 85% of the tokens correctly. The LDA method does not achieve this level of performance.

5SP2. Effects of experimental manipulations of auditory feedback upon vowels. Emily A. Tobey, Dawn Cooper, Heidi Switzer (Commun. Sci. Lab., Dept. of Commun. Disord., LSU Med. Ctr., New Orleans, LA 70112), and Mario Svirsky (MIT, Boston, MA)

The role of auditory feedback upon intermediate vowel production was explored by manipulating the amount and type of information provided by a Nucleus multichannel cochlear implant. Four experimental conditions were examined: (a) two types of control conditions, the implant turned on versus off, and (b) two experimental conditions that varied the channels being stimulated. Twenty repetitions of the vowels [i], [ɛ], [i], [u] in an hVd context were acquired from four adventitiously

deafened adults using multichannel implants. Two of the vowels, [i] and [ɛ], were selected because of previous work indicating significant changes in formant frequencies when the implant was turned on versus off. The vowels [i] and [u] were included as controls since previous work indicated that their formant frequencies failed to change when the implant was turned off. Recordings were digitized at a 20-kHz rate and analyzed using CSpeech. Fundamental frequency, formant frequencies, and duration measures were acquired for each token. Data were statistically treated using a repeated measures ANOVA. Preliminary data indicates that the intermediate vowels rely upon relevant auditory information more heavily for their production than the control, point vowels. [Work supported by NIDCD.]

5SP3. Intelligibility in competing noise by males with partial laryngectomies. Bernice Gerdeman, M. Trudeau, D. Pearl, and R. Wilhelms (Speech and Hearing Sci., Ohio State Univ., 110 Pressey Hall, Columbus, OH 43210)

Although levels of intelligibility among alaryngeal speakers using various speech modes have been investigated [J. G. Clark and J. C. Stemple, *J. Speech Hearing Res.* **25**, 333–338 (1982); M. Kalb and M. Carpenter, *J. Speech Hearing Disord.* **46**, 77–80 (1981)], little data are available for speakers with partial laryngectomies. The specific research questions of the investigation were "To what degree do the patients with hemilaryngectomy or supraglottic laryngectomy maintain intelligibility when speaking in various levels of competing noise?" and "Are there differences in speech intelligibility as a function of whether the partial laryngectomy was in the vertical or horizontal plane?" Nine hemilaryngectomees, nine supraglottic laryngectomees, and nine normal laryngeal male speakers recorded 20 SPIN test sentences with no competing noise and with 75 dB A white noise. Intelligibility scores (identification of each sentence's final word by 30 normal-hearing listeners) were analyzed using logistic regression and indicated that the two surgical groups did not differ significantly from each other in either noise condition and that in both conditions the two groups differed significantly from the normal laryngeal group.

5SP4. Changes in open and speed quotient values as a function of measurement criteria. Christine M. Sapienza, Elaine T. Stathopoulos (State Univ. of New York at Buffalo, Amherst, NY 14260), and Christopher Dromey (Royal Victoria Hospital)

Noninvasive measures of vocal-fold activity have become useful in describing normal and disordered voice production. Specifically, open and speed quotient measures (OQ and SQ) of electroglottographic and inverse filtered airflow waveforms have been used to describe variability in the phases of vocal-fold vibration. Unfortunately, there is little consistency in the criteria used to calculate these quotients. The lack of consistency becomes important when comparing quotient values from similar waveforms (e.g., VFCA waveform) as well as different types of waveforms (e.g., VFCA versus glottal airflow). Four different percentage criteria were chosen to investigate how quotient values change within and across signal types. Twenty percent, 50%, 80%, and 100% of the ac peak-to-peak amplitude of electroglottographic and glottal airflow waveforms were selected. Results indicated different values obtained across criteria levels. Comparison of absolute values from quotient data across criteria or signal type is not recommended. Use of common measurement procedures is needed so that normative data can be established. [Work supported by NIH #DC00516-42.]

5SP5. Minimal spectral contrast for vowel recognition as a function of spectral slope. Andrew Lea and Quentin Summerfield (MRC Inst. of Hearing Res., University Park, Nottingham NG7 2RD, U.K.)

Can listeners locate formants not only from peaks in the envelope of the excitation pattern of a vowel, but also from "shoulders"—features giving rise to zero crossings in the third, but not the first, differential of the excitation pattern—as hypothesized by Assmann and Summerfield [J. Acoust. Soc. Am. **85**, 327–338 (1989)]? Stimuli were steady-state approximations to the vowels /a, i, ɜ, u/ created by summing the first 45 harmonics of 100 Hz. Thirty-nine harmonics had equal amplitudes; the other six formed three pairs that were raised to define three "formants." An adaptive psychophysical procedure determined the minimal difference in level between the 6 harmonics and the remaining 39 at which the vowels were identifiably different from one another. These thresholds were measured through simulated communication channels giving overall slopes to the excitation patterns of the vowels ranging from -1 dB/erb to $+2$ dB/erb. Excitation patterns of the threshold stimuli were computed and the locations of formants were estimated from zero crossings in the first and third differentials. With sloping frequency responses, some formants of some vowels were represented as shoulders rather than peaks, confirming Assmann and Summerfield's hypothesis. Implications for models of formant extraction will be discussed.

5SP6. Age differences in the processing of dynamic acoustic information. Robert A. Fox, Lida G. Wall, and Jeanne Gokcen (Div. of Speech and Hearing Sci., Ohio State Univ., 110 Pressey Hall, Columbus, OH 43210-1002)

The present study examines possible age-related differences in the use of dynamic acoustic information (in the form of formant transitions) to identify CVC words. Sixty-two high-frequency monosyllabic English words were recorded that began or ended in oral or nasal stops. From these words, two sets of tokens were created: an unmodified CVC token representing the whole word and a silent-center version in which approximately 70% of the medial vowel was replaced by silence. Two sets of listeners were required to identify these words in first the silent-center condition and then the whole-word condition. The listeners included 17 college sophomores or juniors and 22 older subjects aged 55–75. All listeners were screened for normal auditory sensitivity. Results demonstrated both age groups' identifications were somewhat better in the whole-word condition than in the silent-center condition. However, the mean correct consonant and vowel identifications of the older listener group was approximately 10% less than that for the younger listeners in the silent-center condition. No age difference was

obtained in the whole-word condition. These data support the hypothesis that older listeners have greater difficulty than younger listeners in processing dynamic acoustic information in the perception of speech. [Supported by NIA Grant #1 R01 AG08353-01.]

5SP7. Evidence for a rhyme and onset model of lexical access in children. W. D. Murphy (Psychology Dept., Univ. of Rochester, Rochester, NY 14627), L. Gerken (SUNY at Buffalo, NY), R. P. Cooper (Virginia Tech., VA), and R. N. Aslin (Univ. of Rochester, NY)

A previous report [Murphy *et al.*, J. Acoust. Soc. Am. Suppl 1 **87**, S73 (1990)] indicated that 4-year olds do not process spoken words and nonwords in a strict left-to-right manner. Instead, in a matching task they appear to search for the longest uninterrupted sequence of segments or to use a rhyming strategy. This poster reports the results of two subsequent studies designed to determine which of these two strategies was employed. The first follow-up study presented children with comparison items that shared the same number of uninterrupted segments as the target word ("little"). Some comparison items rhymed with the target ("gittle"), and others did not ("litten"). Results did not support the uninterrupted-segment matching model but were consistent with the rhyming model. A second follow-up study used monosyllabic stimuli. Results suggested that (1) children treat words with semivowel second syllable consonants as monosyllabic, and (2) both rhyme and similarity of the onset consonant–vowel sequence play a role in lexical access.

5SP8. Effect of consonant–vowel intensity ratio on the intelligibility of spectrally degraded speech. Uma Balakrishnan, Richard L. Freyman, Chiang Yuan Chuan, G. Patrick Nerbonne, and Kelly J. Shea (Dept. of Commun. Disord., Univ. of Massachusetts, Amherst, MA 01003)

Normal-hearing subjects' recognition of spectrally degraded speech was evaluated under conditions where the waveform envelope was modified by altering the consonant–vowel intensity (C–V) ratios. Subjects were required to identify 22 consonants presented in the /a-Consonant-a/ format at suprathreshold and near threshold levels of presentation. In both conditions, the stimuli were processed to limit spectral information. In the suprathreshold condition, stimuli were presented at the natural C–V ratio and at six modified C–V ratios. In the near threshold conditions, three C–V ratios were employed—the natural ratio and two altered C–V ratios. The effect of C–V ratio modification on recognition performance varied across consonants. Within groups of consonants, "high" or "low" C–V ratios predisposed listeners to select certain consonants as the response. For example, for high C–V ratios of the stimuli /afa/ and /asa/, listeners selected /asa/ as the response. For low C–V ratios of these consonants, the predominant response was /afa/. Similar C–V ratio-dependent responses were also obtained for the glide and nasal consonant groups. Finally, for some consonants such as the stops, C–V ratio modification did not affect recognition performance. Results from both studies suggest that in the absence of spectral information, listeners depend on the envelope cues available in the speech waveform for speech sound recognition. [Work supported by the Whitaker Foundation.]

5SP9. The influence of segmental sonority on immediate memory for syllables. Shari R. Speer (Dept. of Psychol., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115) and Aimee M. Surprenant (Yale University, New Haven, CT)

Four experiments demonstrate that the sonority of a syllable's component sounds predicts the presence and magnitude of recency effects in immediate memory. Sonority is determined by co-occurrence restrictions on vowel and consonant sounds within syllable-internal structure. Recency effects were most pronounced in memory for CV strings with

different vowels (high sonority) and least apparent for CV strings with different stop consonants (low sonority). Strings with different nasal consonants and glides (moderate sonority) produced intermediate recency. In addition, lists of high vowel-contrasting syllables (high-moderate sonority) were tested against lists of low vowel-contrasting syllables (high sonority). Overall, results suggest that recall performance for the final positions of an auditory list is predicted by the sonority of the contrast segment. The results are consistent with findings in the categorical perception and auditory memory literatures, and support the notion of a common dimension underlying these well-known effects and the linguistic notion of sonority. The experiments provide additional evidence to support the view that memory for a sound does not differ according to its identity as a vowel or consonant, but instead is influenced by its acoustic properties. [Work supported by NICHD.]

SSP10. Musical duplex perception: Does perceptual dominance reflect general principles or specialized modules? Michael D. Hall and Richard E. Pastore (Dept. of Psychol., SUNY, Binghamton, NY 13901)

In a variant of duplex perception (DP), a phonetic module is claimed to take precedence over nonspeech processing based upon maintained phonetic perception despite discontinued nonspeech perception of the critical stimulus component. Recent attempts to replicate these findings with nonspeech stimuli fail to meet proposed criteria for stimuli used in strong demonstrations of DP. The present musical experiment first established threshold intensities for detecting the sinusoidal, chord-distinguishing note in the context of constant-intensity piano notes. Here, AX chord discrimination followed with distinguishing notes varying in intensity. As with speech, complex perception of chords was maintained at intensities significantly below detection threshold for component perception. Both speech and music findings could demonstrate general principles of perception, with multiple-component stimuli preserving a strong integrative relationship between components more readily perceived as singular events with less stimulus energy than is required for isolating individual components. [Work supported by NSF.]

SSP11. A psychological space for place of articulation. Xiao-Feng Li and Richard E. Pastore (Dept. of Psychol., SUNY, Binghamton, NY 13901)

A psychological space for place of articulation was explored using synthetic /ba/ and /da/ stimuli factorially varying in F2- and F3-onset frequencies. The data from a goodness judgment experiment were subjected to the multidimensional scaling. Within the derived space of each phoneme category, the stimulus centered (or the closest to the center) was designated as the prototype. The results from a speeded classification task indicated that the response time to each phoneme is an ordinal function of the goodness measure. The heterogeneity of the resulting goodness spaces and response times in classification suggests that classification of place of articulation could be the consequence of the subject evaluating the membership function relative to the prototype. [Work supported by NSF.]

SSP12. Interference for "new" versus "similar" vowels in Korean speakers of English. Islay Cowie and Sun-Ah Jun (Dept. of Linguistics, Ohio State Univ., 204 Cunz Hall, Columbus, OH 43210)

Flege (1986, 1987) proposed that although the degree of establishment of a "new" phonetic category in the L2 is proportional to the degree of experience in the L2, equivalence classification prevents adult L2 learners from establishing a phonetic category for "similar" L2 phones. This paper examines the similar English vowels, /i, u, ʌ/, and the new English vowels /ɪ, ʊ/, in productions by Korean-English bilinguals with different degrees of their experience in English and com-

pares with those of English monolinguals. Also, formant values of the Korean vowels were compared between these Korean-English bilinguals and monolingual Koreans (Seoul dialect) to see whether the L2 acquisition affects their Korean production. Preliminary results partially support Flege's hypothesis. Productions of the new English phones were closer to the English norm than that of the similar phone /ʌ/, but only for highly experienced bilinguals. On the other hand, the monolingual norms for the similar phone /i/ are too close to be called merely similar, whereas those for /ʌ/ seem acoustically too different to be categorized as similar. Thus Flege's notion of similar phone needs to be refined or restricted.

SSP13. Perception of prominence in CV sequences by Estonian and English listeners. Ilse Lehist (Dept. of Linguistics, Ohio State Univ., 205 Cunz Hall, Columbus, OH 43210) and Robert Allen Fox (Ohio State Univ., Columbus, OH 43210)

In previous work [I. Lehist and R. A. Fox, *J. Acoust. Soc. Am. Suppl. 1* 87, S72 (1990)], the perception of "prominence" in sequences of noise tokens by native Estonian and American English listeners while independently manipulating individual token duration and amplitude were examined. The present study used the same basic experimental paradigm but with synthetic CVs ([ba]) rather than noise bursts. The basic CV token was 400 ms in duration with 40-ms formant transitions and a 360-ms steady-state vowel. However, in the experimental trials, one token in each sequence could be lengthened to 425, 450, 475, or 500 ms and/or one token (not necessarily the same token) could be increased in amplitude by 3 or 6 dB—these duration and amplitude changes were independent. Listeners were required to indicate which CV in the sequence was "more prominent." Listening tests were given to 33 native speakers of English in Columbus, Ohio and to 40 native speakers of Estonian in Tallinn, Estonia. As in the earlier study, the responses showed that Estonian listeners were more sensitive to token duration in making their "prominence" decisions than were the English listeners. Differences between the two sets of results (obtained using noise versus speech stimuli) will be discussed.

SSP14. Discrimination of synthetic /la-ra/ by birds. Robert J. Dooling, Susan D. Brown (Dept. of Psychol., Univ. of Maryland, College Park, MD 20742), and Catherine T. Best (Wesleyan Univ., Middletown, CT and Haskins Labs., New Haven, CT)

Discrimination latencies were measured for two species of birds tested using pairs of tokens drawn from three synthetic continua based on the /la/ and /ra/ contrast: full-formant syllables, their sinewave-speech analogs, and the critical F3 distinctions as isolated sinewaves. For the full-formant continuum, both species showed a marked improvement in discrimination near the /l-/r/ boundary, whereas for the F3 continuum, neither species showed a peak near the phonetic boundary. These results are comparable to human discrimination of the same continua [Best *et al.*, *Percept. Psychophys.* 45, 237-250 (1989)]. However, for the sinewave-speech continuum, budgerigars showed a performance peak mirroring that for the full-formant syllables, like humans who perceived the sinewave-speech stimuli as speech, while zebra finches showed a linear function mirroring their performance for the F3 sinewaves, like humans who perceived the sinewave speech as non-speech. These data provide new evidence of species similarities and differences in the discrimination of speech and speechlike sounds that strengthen and refine previous findings of sensitivities of the vertebrate auditory system to several acoustic distinctions associated with speech sound categories. [Work supported by NIH Grant DC00198 to RJD.]

SSP15. Adjusting dysarthric speech timing using neural nets. Shirley M. Peters and H. Timothy Bunnell (Speech Processing Lab., Alfred I. duPont Inst., Wilmington, DE 19899)

Speech timing problems associated with dysarthria often involve the presence of periods of extraneous silence and nonspeech sounds as well as inappropriately timed or misplaced speech gestures. This study evaluated the performance of neural networks in detecting the presence of inappropriate or nonspeech sounds and extraneous silence. The "opt" neural network program [E. Barnard and R. Cole, OGC Tech. Rep. No. CSE 89-014] that uses a conjugate gradient algorithm to adjust node weights was trained to recognize breaths and silence in a reading of the rainbow passage by a single dysarthric (Cerebral Palsy) talker. Input to the network consisted of a sequence of frames of parameters derived from spectral analysis of the speech. The output was a binary (speech/nonspeech) decision for the segment of signal corresponding to the middle frame of the input sequence. Networks of various size and configuration were trained on half the available data and tested on the remaining data. The best network configurations correctly identified approximately 99% of the frames in the training set and about 97% of the frames in test datasets.

SSP16. Analog speech processor based on the auditory periphery. Weimin Liu, Andreas G. Andreou, and Moise H. Goldstein, Jr. (Sensory Commun. Lab., Elec. and Comp. Eng. Dept., Johns Hopkins Univ., Baltimore, MD 21218)

Neurophysiological studies of the auditory periphery show that synchrony coding is a robust representation of speech signals. Recent research demonstrated that models of the auditory periphery give superior performance as the front-end signal processor of a speech recognition system, especially when speech inputs are corrupted by noise [H. M. Meng and V. W. Zue, International Conference on Spoken Language Processing, 1053-1056 (1990)]. Here, an auditory periphery model is realized on silicon with analog integrated semiconductor technology to provide a real-time, low-power dissipation preprocessor for speech processing tasks such as speech recognition and aids for the deaf. The model, including the middle ear, the basilar membrane and the hair cell, and synapses, is presented along with the design features of analog CMOS implementation. The pattern of the multichannel outputs resembles the neurogram of the auditory-nerve fibers, i.e., the time-varying instantaneous discharge rates in fibers of various characteristic frequencies, in response to speech signals such as consonant-vowel syllables [H. E. Secker-Walker and C. L. Searle, J. Acoust. Soc. Am. **88**, 1427-1436 (1990)].

SSP17. The analysis of F_0 reset in relation to phrase dependency structure. Yoshinori Sagisaka (ATR Interpreting Telephony Res. Labs., Japan)

In Japanese speech, the reset of phrasal F_0 downtrends has mainly been analyzed with respect to the number of prosodic units (lg) between the phrase preceding the boundary and the phrase it directly qualifies. This parameter reflects the local structure of the sentence following the boundary and corresponds to the forward limit of local F_0 control. In the analysis, an additional parameter (ld) that reflects the left local structure of the sentence is introduced. This parameter is the number of units that modifies the phrase preceding the boundary. To measure the F_0 reset, the ratio (F_{0r}) of the averaged F_0 values of the phrases preceding and following the boundary is used. Here, F_0 reset at right-branching boundaries is expressed as " $F_{0r} > 1$ when $lg > 1$." Using these two parameters lg , ld and the average F_0 ratio F_{0r} , F_0 resetting phenomena were analyzed quantitatively at about 2000 phrase boundaries in 500 sentences. The results show that (1) F_{0r} increases in proportion to lg , (2) at right-branching boundaries ($lg > 2$), F_{0r} increases in proportion to ld , and (3) at left-branching boundaries, there is no strong correlation between F_{0r} and ld , and F_{0r} is greater than 1 only at clause boundaries or when a following unit is a headlike one. Moreover, it is also observed that F_{0r} is larger at coordinate phrase boundaries than at other boundaries. These facts support the usefulness of the new parameter ld and will contribute to the quantitative treatment of F_0 control for speech synthesis.

SSP18. Use of stress in an isolated word recognizer for French. Kathleen M. Bishop (The Inst. for Res. in Cognitive Sci., Univ. of Pennsylvania, 3401 Walnut St., Ste. 420C, Philadelphia, PA 19104 and Dragon Systems, Inc., 90 Bridge St., Newton, MA 02158) and Caroline B. Huang (Dragon Systems, Inc., Newton, MA 02158)

An isolated word recognition system for French is now being developed. This effort is part of an ongoing collaborative project between Dragon Systems, Inc. and Lernout and Hauspie Speechproducts n.v. to port and adapt Dragon's large vocabulary speech recognition system for American English to five European languages: French, Spanish, German, Italian, and Dutch. In English, unstressed syllables have been observed to be quite reduced compared to stressed syllables, while unstressed syllables in French have not been observed to be greatly reduced compared to their stressed counterparts. In the American English recognition system, performance is enhanced when the syllable stress is taken into account. Whether taking stress into account will enhance performance of the French recognizer is now being investigated. The preliminary French system recognizes 1000 words and is speaker adaptable. In developing the system, 3000 tokens from one native speaker of French have been collected. The capabilities of the recognizer will continue to expand and the collection of data will continue to take place. Preliminary results from the acoustic analysis of the speech data and from the experiments with the recognizer will be discussed. [This work is supported by a joint program between Dragon Systems, Inc. and Lernout and Hauspie Speechproducts n.v.]

SSP19. Speaker-independent speech recognition with word models generated from written text. E. L. Bocchieri and J. G. Wilpon (AT&T Bell Labs., Rm. 2C-543, 600 Mountain Ave., Murray Hill, NJ 07974)

Most current subword-based recognition systems require manual generation of the lexical transcription of vocabulary words. In addition, they generally use application specific data to train the acoustic subword models. Hence, a new database is collected for every task. This study shows that the generation of application specific word models can be *completely automated* by combining application-independent phonetically based subword models according to a lexicon provided by a text-to-phone transcriber. Continuous-density hidden Markov models of a set of phonetic units have been trained with the segmental K -means parameter estimation algorithm for both the TIMIT and the DARPA Naval Resource Management speech corpora. Each new (application specific) vocabulary word is typed into a text-to-phone converter [M. D. Riley, Proc. ICASSP, 1991 (to appear)] that provides different transcriptions ranked in order of likelihood. The most likely transcription is used to combine phone models into a word model. System evaluation is performed on a 147-item vocabulary (e.g., digits, numerals, words, and two-word phrases) chosen independently from the training data and collected from ten speakers in an office environment (i.e., a different environment than was used to record the training data). The experimental evaluation made use of initial and final silence models, first- and second-order energy and cepstrum derivatives, and state duration models. Currently, the recognition accuracy on the entire vocabulary is 90.0% and 97.8% on a 20-word subset (of 2-4 syllable words). These accuracies are consistent across the two different training corpora.

SSP20. Loudness levels of three complex stimuli and model predictions. Patricia S. Jeng¹ (Ctr. for Res. in Speech and Hearing Sci., City Univ. of New York, New York, NY 10036), Joseph L. Hall (Acoust. Res. Dept., AT&T Bell Labs., Murray Hill, NJ), and Harry Levitt (City Univ. of New York, New York, NY 10036)

Loudness levels were measured for three test stimuli (speech, narrow-band noise, and square wave) at three levels in three test conditions (test stimulus in quiet, test stimulus in presence of masker, and total loudness of test stimulus plus masker). Loudness levels were measured in the traditional way by matching loudness of the test stimulus to that of a 1-kHz tone. In addition, loudness levels were measured using

a narrow-band noise as the reference stimulus. The bandwidth of the narrow-band noise used as test stimulus (NBN-150) was 150 Hz, and the bandwidth of the narrow-band reference stimulus (NBN-120) was 120 Hz. Both stimuli were arithmetically centered at 1 kHz. The loudness level functions for speech and for NBN-150 have slopes of approximately unity. However, the loudness level function for the square wave has a slope of greater than unity below about 50 dB SL and the slope decreases above about 50 dB SL. Loudness predictions of two models are computed and compared to the experimental measurements.^{a)}Work done as a consultant at AT&T Bell Laboratories, Acoustics Research Department.

5SP21. Human frequency following responses to vowel-like stimuli. A. K. Ananthanarayan (Dept. of Audiology and Speech Pathology, Univ. of Tennessee, 457 S. Stadium Hall, Knoxville, TN 37996) and H. S. Gopal (Univ. of California, Santa Barbara, CA)

The frequency following response (FFR) reflects sustained neural activity within the brain stem, phase locked to the cycles of the stimulus waveform. Auditory-nerve single unit responses have been shown to utilize the property of phase locking to represent the stimulus spectrum of steady-state vowels. The purpose of this study was to determine if the scalp-recorded FFR would reveal similar neural representation of steady-state vowels. Scalp-recorded FFRs were obtained from six normal-hearing subjects in response to five different synthesized steady-state vowels [a/, /i/, /u/, /æ/, and /o/] presented binaurally at 70, 50, and 30 dB nHL. Spectral analyses of the FFRs indicated that, for each vowel, phase locking is robust at the fundamental frequency and the first three harmonics with phase locking to the fundamental preserved even at stimulus levels close to response threshold. Spectral peaks corresponding to the first two formants were observed at only the higher intensities. These results, while not entirely consistent with the single-unit data, seem to suggest that the FFRs reflect neural activity related to the processing of low pitch.

5SP22. A phase cancellation method for tinnitus. M. D. Judd and A. Ünäl (RTS, 3100 Central Expressway, Santa Clara, CA 95051)

Tinnitus is an intrinsic feedback noise generated within the inner ear. Objective tinnitus is the kind in which a sound field does exist in the outer ear canal. Recent findings on the objective tinnitus problem strongly suggest that the signal(s) emanate from damaged sensor hairs inside the cochlea. The ringing, or noise, that the patient "hears" is most likely the firing of neighbor sensor hairs, due to the damaged sensor. Thus these neighbors comprise some signal bandwidth over which the patient detects the ringing. In the greater number of patients, this phenomenon is diagnosed as due to a single tone. Current signal methods used to mask the tone, or noise, emanating from the cochlea typically employ wide bandwidth signals that overlap more than the full bandwidth of the tone (or noise). These methods inherently create noise in themselves and do little to remove the signals being generated from the inner ear. This paper presents a simple viable method to cancel a single tone, in the local region of the damage sensor, in the cochlea.

5SP23. Auditory-contrast sensitivity in normal and hearing-impaired listeners. David A. Hillier and James D. Miller (Central Inst. for the Deaf, 818 S. Euclid Ave., St. Louis, MO 63110)

The peak-to-valley ratio (depth of modulation in dB) of sinusoidal spectral envelopes necessary to distinguish such modulated spectra from flat spectra was investigated in several experiments. In addition, several adaptation experiments were performed to test the hypothesis of envelope frequency channels in the auditory system similar to spatial frequency channels in the visual system. Factors in hearing impairment that lead to contrast enhancement (recruitment) and contrast reduction (loudness saturation and loss of spectral resolving power) are discussed. Finally, auditory contrast sensitivity functions for two hearing-impaired subjects are presented.

5SP24. An automated test battery for assessing the auditory fitness of aviators. Gerald B. Thomas, Carl E. Williams, and Andrew H. McCordie (Sensory Sci. Div., Naval Aerospace Med. Res. Lab., Pensacola, FL 32508-5700)

A microcomputer-based battery of tests has been developed to assess the performance of aviators on auditory tasks deemed important in the operation of aircraft. Included in the test battery are measures of hearing threshold levels, speech perception in cockpit noise, speeded response to auditory signals, auditory attention management, and auditory short-term memory. All auditory stimuli are computer synthesized or are digitized from analog sources, and all test administration and scoring procedures are automated; test duration is approximately 30 min. The test battery was administered to six aviator and nonaviator populations ($N = 120$) and showed significant ($p < 0.05$) performance differences among the populations on one or more of the test battery elements. Test-retest reliabilities ranged from 0.71 to 0.88 ($n = 64$) within a restricted variance population (i.e., student naval aviators). Correlations between test battery scores and simulated "real world" performance are currently being determined.

5SP25. The effects of noise on the measurement of the $2f_1 - f_2$ otoacoustic emission. Richard W. Harrell, Lawrence L. Feth (Div. of Speech and Hearing Sci., Ohio State Univ., Columbus, OH 43210), and William Melnick (Ohio State Univ., Columbus, OH 43210)

The effects of noise exposure on the measurement of the cubic distortion product emission ($2f_1 - f_2$) were measured. TTS was induced using white noise presented at four intensities, each for three different time periods, giving a total of twelve conditions. The subjects were normal-hearing adults. The trial conditions were randomized, with an appropriate rest period between each condition. The amount of temporary threshold shift (TTS) in dB was obtained by comparing pre- and post-noise pure-tone thresholds for the 500-Hz to 8-kHz range by octave. The level of $2f_1 - f_2$ was measured prior to and immediately following the threshold testing. Changes in the distortion product emission were determined by comparing pre- and post-noise emission measures. Preliminary results suggest that noise exposure may reduce the level of the distortion product emission. The degree to which this effect was seen appeared to be related to both intensity and duration of the noise exposure.

5SP26. Importance and generality of various sound assessment criteria. I. Naive listeners. Kim M. Smith and Tomasz Letowski (Dept. of Commun. Disord., Penn State Univ., 5 Moore Bldg., University Park, PA 16802)

A series of criteria and their definitions has been developed describing auditory image attributes that are felt to be important in sound quality assessment. These criteria were presented to two matching groups of 25 naive young listeners. Group I read the definitions for each criterion and then rated the attributes on two, five-point scales: (1) an importance scale and (2) a generality scale. Group II rated each criterion on the same scales, but they did not read any definitions. The same 50 listeners were retested under identical conditions. The following results will be discussed: (1) importance and generality of various criteria, (2) differences between groups, (3) differences between test and retest data, and (4) application of results for future research.

5SP27. Auditory processing of complex signals using the multichannel EWAIF. Jayanth Anantharaman (Dept. of Electrical Eng., Ohio State Univ., Columbus, OH 43210), Lawrence L. Feth,

The envelope weighted average instantaneous frequency (EWAIF) has been used to model listener performance in complex signal discrimination [L. L. Feth, *J. Acoust. Soc. Am. Suppl.* 1 88, S48 (1990)]. Previous work with EWAIF has involved narrow-band signals that are confined entirely to a single critical band; consequently, it is sufficient to compute a single EWAIF value from the signal. Here, the application of EWAIF to wideband signals is extended. The signal is first passed through the gamma-tone filter bank of the Patterson-Holdsworth auditory sensation processing model [R. Patterson, *J. Acoust. Soc. Am. Suppl.* 1 88, S26 (1990)] and an EWAIF is computed from the output of each channel. This leads to a vector of EWAIF values, which are used in two ways: (1) the EWAIF values from the different channels are combined back into a composite EWAIF value, which is then used to model listener performance in complex sound perception such as profile analysis, comodulation masking release, and modulation masking; and (2) the vector of EWAIF values is used as a feature vector in a vowel recognition task. Preliminary results will be presented and further modifications to the model discussed. [Work supported by a grant from AFOSR.]

5SP28. A new technique for measuring spectral shape discrimination. Gerald Kidd, Jr. and Christine R. Mason (Dept. of Commun. Disord., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215)

A new technique for measuring the ability of listeners to discriminate between sounds on the basis of spectral shape—called “auditory profile analysis”—is described. In this technique, the signal is a series of intensity increments and decrements to an equal-level multitone reference spectrum. For any stimulus trial, the signal is either an intensity increment to the odd-numbered tones and an intensity decrement to the even-numbered tones, or vice-versa. The advantage of the technique is that it reduces, by one-half (in a two-down, one-up adaptive procedure), the range of the random, within-trial rise in overall level needed to limit the usefulness of level cues. Using this technique, discrimination performance was measured for a group of normally hearing listeners for broadband, low-pass, and high-pass filtered conditions, and for individual hearing-impaired listeners. The results indicated that the new technique appears to be well suited for studying auditory profile analysis in hearing-impaired listeners where the range of sound intensities that may be presented is often quite limited. [Work supported by NIH/NIDCD.]

5SP29. Detection of silent temporal gaps in narrow-band noise stimuli having second-formantlike properties of voiceless stop/vowel combinations. C. Formby (Dept. of Otolaryngol., Johns Hopkins Univ. School of Medicine, Carnegie 442, Baltimore, MD 21205), C. Barker (Univ. of Florida, Gainesville, FL 32611), H. Abbey (Johns Hopkins Univ., Baltimore, MD 21205), and J. Raney (Univ. of Massachusetts, Amherst, MA 01003)

Delattre *et al.* [*J. Acoust. Soc. Am.* 27, 769–773 (1955)] reported that good-quality stop consonant/vowel pairs could be synthesized in two-formant representation by replacing the initial part of the second-formant (F_2) transition with a silent temporal gap. The best silent gap for each stop varied with the timing and extent of the F_2 transition, which we know is (1) dependent on coarticulation between the stop and vowel and (2) represented acoustically by the difference or change in frequency from the stop burst to F_2 of the following vowel. Motivated by their report, temporal gap detection thresholds were measured in narrow-band noise-burst markers having acoustic characteristics that simulated isolated steady-state F_2 properties of voiceless stop /p, t, k/ and vowel /i, æ, u, ɔ/ combinations. Gap detection thresholds increased systematically as a function of the difference between the simulated stop and vowel F_2 frequencies, which accounted for 76% of the variance in the results. Differences in other stimulus features had little influence on detection. [Research supported by NIH.]

5SP30. Gap thresholds obtained with auditory steady-state responses. Craig A. Champlin (Dept. of Speech Commun., Univ. of Texas, Austin, TX 78712)

Human gap thresholds were obtained electrophysiologically using auditory steady-state responses (ASRs). Three types of stimuli—bandpass, low-pass, and high-pass noise—were presented at three sensation levels (30, 50, and 70 dB SL). The cutoff frequency for the latter two noises was 1000 Hz. The otherwise-continuous stimuli were interrupted 40 times/s, thus creating a repetitive sequence of gaps. The duration of the gap was 0, 2, 4, 6, 8, 10, 15, 20, or 24 ms. The magnitude and phase of the time-averaged ASRs were obtained for each stimulus condition. Gap thresholds obtained with the bandpass and high-pass noises were similar, while thresholds obtained with the low-pass noise were larger. These results are consistent with existing psychophysical data.

5SP31. Effects of temporal-envelope correlation on the strength of vertical fusion of noise-band pairs. P. W. Rappold, L. L. Mendoza, and M. J. Collins (Dept. of Commun. Disord., 163 M&DA Bldg., Louisiana State Univ., Baton Rouge, LA 70803)

Vertical fusion is the perceptual grouping of co-occurring auditory signals into a single acoustic event. In contrast, *horizontal streaming* is the perceptual grouping of auditory signals occurring in a temporal sequence. In the present experiment, “strength” of fusion was inferred from the interstimulus interval (ISI) necessary to evoke a predominant percept of horizontal streaming rather than vertical fusion. Perceptual grouping heuristics suggest “strength” of fusion is greater for AM noise bands with correlated temporal envelopes than for AM noise bands with uncorrelated temporal envelopes. It was therefore predicted that a shorter ISI would be necessary to shift the percept from vertical fusion to horizontal streaming when noise-band envelopes were correlated than when they were uncorrelated. This prediction proved accurate for two of four subjects. The remaining two subjects, however, showed no significant difference between the ISIs needed to shift the percept from fusion to streaming. The results will be discussed in reference to implications for the bases of the comodulation masking release phenomenon.

5SP32. Restarting the adapted binaural system in a background of noise. Ervin R. Hafer and Kourosh Saberi (Dept. of Psychol., Univ. of California, Berkeley, CA 94720)

The rate-dependent reduced binaural sensitivity to interaural information coming after the stimulus onset has been called “binaural adaptation.” It has been shown that an appropriate restarting trigger presented during the ongoing stimulus can end the adaptation, allowing for lateralization based on information beyond the restart. Effective triggers include short bursts of noise and gaps in the stimulus. It has been reasoned that continuous spectral changes might constantly restart the system and thus preclude adaptation. To test this, click trains have been presented for lateralization in a continuous background noise and with click rates sufficiently high to produce adaptation in the quiet. Our hypothesis was wrong. While binaural thresholds increased with noise level as the noise spectrum levels rose from 0 to 20 dB, no change was observed between the *forms* of the functions obtained in quiet and in noise. Similarly, the inclusion of a gap in the center of the train triggered a restart both in the quiet and in the noise background. Thus a spread of energy into quiet bands as has been postulated in the past as an explanation for restarting is insufficient.

5SP33. Aging and discrimination of interaural time differences. Daniel Ashmead, Marta Tetzeli, David Chandler, and Amy Thomas (Dept. of Psychol., Vanderbilt Univ., Nashville, TN 37240)

Discrimination of interaural time differences (ITDs) was compared between ten younger adults (18–24 years) and ten older adults (66–80

years, with normal or near-normal hearing). Both the mean thresholds and standard deviations for older adults were elevated approximately fourfold compared to the younger subjects. The age difference is consistent with previous reports of age-related loss in sensitivity to ITDs. However, in our study and in previous studies, results may have been confounded by a methodological factor. The older adults had very large excursions across stimulus levels in the adaptive (staircase) psychophysical procedure used. Thus the age difference may reflect attentional as well as auditory factors. Further research using the method of constant stimuli in a small number of older adults is in progress. Preliminary results indicate that there is still an age difference in interaural processing, but that the difference is smaller than methods using the adaptive procedure have indicated. The findings have implications not only for age-related changes in interaural processing, but also for the use of psychophysical testing methods with older adults.

5SP34. Human audition of ultrasound. Martin L. Lenhardt, Alexander M. Clarke, Ruth A. Skellett, David A. Soltanpour, and Alfred L. Ochs (Med. College of Virginia, Virginia Commonwealth Univ., Box 168, Richmond, VA 23298-0168)

Tones above 20 000 Hz are not inaudible to humans as the term "ultrasound" implies, but rather are audible only via bone conduction. Placing a high-frequency vibrator on the skin over the head, neck, or upper trunk results in a high-pitched auditory percept, yielding no sense of cutaneous stimulation. Audibility thresholds for tones of 25 and 62.5 kHz were determined in four normal-hearing subjects for various sites on the upper torso. The results of these ultrasonic "maps" indicate that the area of the mastoid bone is the most sensitive. Position and orientation of the vibrator on the mastoid influence sensitivity. Small changes of frequency and/or vibrator orientation affect the "ear" in which the sound is heard. This finding is in contrast to a previous report from this

laboratory of 60- to 80-dB attenuation between mastoid bones in a dry skull [Dunlap *et al.*, *Otolaryngol. Head Neck Surg.* **99**, 389-391, (1988)]. By placing water in the cranial vault, skull attenuation of 17 dB (25 kHz) and 36 dB (62.5 kHz) was obtained, values better matching the perceptual findings. It is hypothesized that standing waves are created intracranially, which in part determine the ear of perception. It is possible that cochlea inner hair cell stereocilia and/or vestibular type I cells in the striolar region of the saccule are set into ultrasonic resonance [Strelhoff *et al.*, *Hear. Res.* **18**, 169-175 (1985)], inducing a different form of stimulation than traditional air conduction hearing. [Work supported by Hearing Innovations, Inc., Tucson, AZ.]

5SP35. Gender classification based upon statistics of the spectral distributions: Walking perception. Xiao-Feng Li, Richard E. Pastore (Dept. of Psychol., SUNY, Binghamton, NY 13901), and Robert J. Logan (IBM, Poughkeepsie, NY 12602)

The current study continued the previous research using as stimuli the sounds of humans walking on a hard surface [Li *et al.*, *J. Acoust. Soc. Am. Suppl.* **1** 87, S24 (1990)] to investigate the acoustic properties that delineated categories of ecologically valid auditory source events. Various statistics of the walking spectra were computed, and subjected to a factor analysis in order to identify the spectral properties that differentiated between actual gender class of source events (male and female walkers). Two classes of information were identified as important in distinguishing between the male and female spectra: (1) central tendency and (2) slopes of spectral attack and decay. A multiple regression analysis indicated the importance of these classes of information in determining the perception of the walker gender. A follow-up experiment then manipulated these statistical properties to verify their contributions in the perceptual classification of walker gender. [Work supported by NSF.]

WEDNESDAY AFTERNOON, 1 MAY 1991

LIBERTY B, 12:40 TO 3:15 P.M.

Session 5UW

Underwater Acoustics: Scattering and Reverberation

Stanley Chin-Bing, Chair
Naval Ocean Research and Development Activity, Code 220, Numerical Modeling Division,
Stennis Space Center, Mississippi 39529-5004

Chair's Introduction—12:40

Contributed Papers

12:45

5UW1. Generalized Fresnel corrections for scattering in underwater sound channels with rough boundaries. Suzanne T. McDaniel (Appl. Res. Lab., Penn State Univ., University Park, PA 16804)

The Fresnel corrected Kirchhoff approximation for forward scattering from rough surfaces is extended to treat the general case of boundary scattering in a refractive underwater sound channel where multiple interactions with the surface may occur. Expressions are obtained for the reflected intensity and the spatial covariance of the scattered acoustic field in the high-frequency limit. Numerical examples are presented

to demonstrate the strong predicted dependence of both the reflectivity and spatial covariance on refraction. [Research supported by ONT with technical management provided by NCSC.]

1:00

5UW2. Far-field considerations in surface and bottom scattering. Darrell R. Jackson and Eric I. Thorsos (Appl. Physics Lab., College of Ocean and Fishery Sci., Univ. of Washington, Seattle, WA 98105)

It is sometimes argued that the concept of scattering cross section is

inapplicable for most surface and bottom scattering measurements, because the acoustic receiver is invariably in the near-field zone of the ensonified patch and scattering cross section is defined in the far-field limit. Winebrenner and Ishimaru [IEEE Trans. Antennas Propag. AP-34, 847-849 (1986)] have shown that the correlation length of the surface field defines the relevant far-field region for stochastic scattering. This sets a less stringent constraint than the far-field criterion based on the ensonified patch size and greatly extends the domain of applicability of the scattering cross-section concept. Using a standard formalism that includes near-field phase terms, it is shown how the scattering cross section can be used even in the near field of the ensonified patch. The result is a proof of the method commonly used in reverberation calculations: One integrates over the ensonified patch with the appropriate propagation loss factors and bistatic scattering cross section as integration variables. This assumes that the far-field criterion based on the correlation length is satisfied. [Work supported by ONR.]

1:15

5UW3. Short-range surface and bottom backscattering strength measurements from the CEAREX 89 arctic experiment. Thomas J. Hayward and T. C. Yang (Naval Res. Lab., Code 5123, Washington, DC 20375)

Short-range SUS reverberation data collected during the CEAREX 89 experiment in the Norwegian/Greenland Sea are analyzed to extract under-ice and bottom backscattering strengths from direct-path measurements. Because of the mildly bistatic experimental geometry, propagation paths with different grazing angles contribute to the reverberation at the same time; therefore, a least-squares method is used to recover the angular dependence of backscattering strength. Ice backscattering strengths are compared with the results of Milne, while bottom backscattering strengths are compared with results for similar ocean bottoms. [Work supported by ONR arctic sciences program.]

1:30

5UW4. Sea surface scattering using 2-D perturbation theory. Eric I. Thorsos (Appl. Physics Lab., Univ. of Washington, Seattle, WA 98105)

Comparisons with exact integral equation results for 1-D surfaces have shown that perturbation theory, when carried to fourth order in kh for the scattering cross section, gives accurate predictions at low frequencies for low-grazing-angle backscattering from the "classical" sea surface. Here, k is the acoustic wave number, h is the rms surface height, and the classical sea surface assumes linear gravity waves with no bubbles. It will be shown that 1-D perturbation theory is accurate for frequencies up to 800 Hz for a Pierson-Moskowitz surface spectrum with a wind speed of 20 m/s (39 kn). For this example, $kh = 7.1$. Perturbation theory has now been extended to 2-D surfaces, in other words, to surfaces that vary in two horizontal directions. Similar accuracy for perturbation theory can be expected for 2-D and 1-D predictions. Examples of 2-D results for scattering strengths will be presented. [Work supported by ONR.]

1:45

5UW5. The effects on long-range ocean acoustic propagation of step-periodic roughness along the interface of a shear-supporting basement. Stanley A. Chin-Bing (Naval Oceanographic and Atmospheric Res. Lab., Stennis Space Ctr., MS 39529) and Joseph E. Murphy (Univ. of New Orleans, LA 70148)

It has been observed [Evans and Gilbert, J. Acoust. Soc. Am. 77, 983-988 (1985)] that one of the effects of ocean subbottom roughness is to slightly decrease the acoustic propagation in the ocean waveguide. While this transmission loss is small, it is cumulative and the net effect over a long range can be a significant decrease in the received acoustic signal. To demonstrate this cumulative loss, Evans and Gilbert used a

full-wave range-dependent coupled mode model (COUPLE) to account for backscatter from a periodic-step subbottom. COUPLE represented a significant advancement in ocean acoustic propagation modeling since it could correctly include the coupled backscattered acoustic field due to range variations in the ocean boundaries. Their roughness simulation has been duplicated using our full-wave range-dependent finite-element ocean acoustic models (FOAM, FFRAME, and PE-FFRAME) and also included the effects of ocean subbottom compression-shear conversion using this full-wave range-dependent finite-element ocean seismo-acoustic model (SAFE). Using a cw full-field backscatter method, it has been possible to isolate the various effects due to the shear and roughness to give insight into the complicated processes that account for cumulative losses over long ranges. [Work supported by ONR/NOARL.]

2:00

5UW6. Acoustic scattering in a three-dimensional oceanic waveguide using boundary integral equation methods. Trevor W. Dawson (Defence Res. Establishment Pacific, F.M.O., Victoria, B.C. V0S 1B0, Canada)

This paper develops in more detail the theory for a fully three-dimensional version of a recently published [T. W. Dawson and J. A. Fawcett, J. Acoust. Soc. Am. 57, 1110-1125 (1990)] boundary integral equation method (BIEM) formulation, for the computation of the scattering of underwater sound from compact deformations of an oceanic waveguide's surfaces. The method allows for three-dimensional sources and boundary deformations in an otherwise uniform waveguide. The technique involves only integrations over the compact surface of the waveguide deformation. The implementation is illustrated for a seamountlike deformation in an oceanic waveguide.

2:15

5UW7. Scattering matrix and boundary integral equation methods for long-range propagation in an acoustic waveguide with repeated boundary deformations. Trevor W. Dawson (Defence Res. Establishment Pacific, F.M.O., Victoria, B.C. V0S 1B0, Canada)

Boundary integral equation methods (BIEM) provide an accurate model for predicting the scattering of acoustic radiation from compact deformations of the walls of a waveguide. In numerical implementation, however, the denseness of the resulting coefficient matrix inhibits the modeling of scattering over very extensive deformations. It is shown in this paper how scattering matrix methods can be combined with BIEM to model long-range propagation in a two-dimensional range-dependent waveguide. The extension to three-dimensional sources is also indicated.

2:30

5UW8. Scattered field calculations for three-dimensional fluid-elastic interfaces. Kevin LePage and Henrik Schmidt (Dept. of Ocean Eng., MIT, Rm. 5-007, Cambridge, MA 02139)

The calculation of the three-dimensional scattered field caused by the insonification of two-dimensionally rough interfaces in horizontally stratified elastic media is discussed. In general, the scattered field may be calculated either deterministically for a *realization* of surface roughness, or stochastically as scattered covariances for a surface with a specified roughness power spectra. The perturbation method of Kuperman and Schmidt [J. Acoust. Soc. Am. 86, 1511-1522 (1989)] is used in the full three-dimensional formulation to model an isotropically rough elastic ice plate floating over a fluid half-space. Scattered field realizations are calculated for the point source configuration, exhibiting the antiplane shear (*SH*) coupling unique to the 3-D model. The ability of explosive sources in the water column to generate *SH* waves in ice has been observed experimentally and it is argued that rough surface scattering is the likely mechanism that facilitates this mode conversion. [Work supported by ONR.]

5UW9. Directional decomposition of the acoustic field scattered from rough ice. J. Robert Fricke (MIT/WHOI Joint Prog., Bigelow 415, Woods Hole Oceanographic Inst., Woods Hole, MA 02543) and Arthur B. Baggeroer (MIT, Cambridge, MA 02139)

At low frequency, ≈ 50 Hz, the acoustic field scattered from a single ice roughness element such as an ice keel or an ice edge contains plane-wave components in virtually all directions. The scatterer affects long-range propagation in two ways: It draws energy out of the coherent field, thus causing attenuation, and it injects incoherent energy into the environment, thus raising the noise level. In general, these two phenomena occur in different directions. The coherent energy deficit occurs in the specular direction and the incoherent scatter occurs over a wide band of angles. Using data generated with a finite-difference model, these phenomena are demonstrated. A plane-wave energy flux calculation reveals quantitatively how much energy is scattered in each direction. Every scattering element has a characteristic directional spectrum, thus implying a characteristic effect on long-range propagation. Results for an ice edge and several keels are compared. An ice edge acts primarily to attenuate the coherent field while the keels both attenuate the coherent field and scatter incoherent energy. In all cases, the energy deficit in the coherent field is greater than the incoherent scattered energy. The difference is converted to plate wave energy in the ice. For grazing angles around 7° , the coherent energy reflection coefficient R^2 is

about 0.97 for an ice edge. For keels, R^2 varies from ≈ 0.95 down to ≈ 0.65 , depending on keel size.

3:00

5UW10. Basic formulation for the simulation of underwater reverberation. Yasushi Sudo (Graduate Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

A basic equation to simulate underwater reverberation for monostatic sonar is obtained. The derivation is based on these simple assumptions: The phase shift caused by scatterers has a uniform distribution, the scatterers are independent of each other, the reverberation is approximated as a quasistationary Gaussian process, the bandwidth of the transmit signal is narrow, and the sound propagation speed is constant. Using these assumptions, the stochastic property of a modified DDM (Doppler density matrix) is examined, which is an extension to the original DDM of REVGEN [D. W. Princehouse, tr7511, Appl. Phys. Lab., Univ. of Washington (1975)]. The correlation of the modified DDM has a simple form. Using the modified DDM, the reverberation is expressed as an overlapped summation of the randomly modulated convolution of the transmit signal and white noise of finite duration. The modulation is related to the Doppler shift caused by the motion of the sonar platform and the scatterers. Using the formulation, an extension to the many beams case is also examined.

WEDNESDAY AFTERNOON, 1 MAY 1991

LIBERTY A & B, 4:45 TO 5:45 P.M.

Plenary Session

Alan Powell, Chair
President, Acoustical Society of America

Presentation of Awards

R. Bruce Lindsay Award to Yves H. Berthelot

R. Bruce Lindsay Award to Joseph M. Cuschieri

Gold Medal to Manfred R. Schroeder

Session 6EA

Engineering Acoustics: Measurements and Instrumentation

Allan J. Zuckerwar, Cochair

NASA Langley Research Center, Mail Stop 238, Hampton, Virginia 23665

Kenneth D. Rolt, Cochair

Ocean Engineering Department, Massachusetts Institute of Technology, Cambridge, Massachusetts 02139

Chair's Introduction—8:45

Contributed Papers

8:50

6EA1. Improvements to the resonance apparatus. Gilbert F. Lee, Brian A. Miller (Naval Surface Warfare Ctr., Silver Spring, MD 20903), and David H. Hanson (Mare Island Naval Shipyard, Vallejo, CA 94592)

The resonance apparatus has been used to determine the complex Young's modulus of polymeric materials. A bar of material is excited into resonance. One end of the test bar is adhesively bonded to a mechanical shaker via a mounting block, while the other end of the test bar is bonded to an accelerometer. A second accelerometer is bonded to the shaker mounting block. At resonance, the complex Young's modulus is determined at four to six discrete frequencies over 1.5 decades of frequency. There are two disadvantages to the above technique. The first disadvantage deals with accelerometer leads. Accelerometer leads are fragile and are easily broken off. Also the accelerometer lead on the free end of the test bar is an unknown added end mass. In addition, this lead may also be an unknown source of absorption. The second disadvantage is that the complex modulus is only determined at resonance. The goal of this presentation is to improve the resonance apparatus by replacing the accelerometers with a noncontacting magnetic pickup and to determine the complex modulus off resonance, which extends the usable frequency range to about 3 decades.

9:05

6EA2. Noninvasive measurement of dynamic pressure inside pipes using piezoelectric strip. Gerard P. Carroll (David Taylor Res. Ctr., Bethesda, MD 20084-5000)

This paper presents the use of piezoelectric material wrapped around the outside diameter of a pipe wall for measuring the dynamic pressure inside the pipe. Recently available piezoelectric materials such as polyvinylidene fluoride (PVDF) are flexible enough to wrap around a pipe wall and provide an excellent transducer for measuring the breathing response of the pipe wall to the internal dynamic pressure. The strain induced in the piezoelectric element associated with radial deformation of the pipe wall is measured. It is shown that the sensor, since it sums the response around the entire circumference, will discriminate against strains associated with bending of the pipe for all circumferential orders which occur below the ring frequency. The relationship between the measured piezoelectric element voltage and pressure, taking into account the pipe wall characteristics, is presented. Various configurations of the device are discussed and experimental results comparing the sensor output to that of a flush mounted hydrophone are presented.

9:20

6EA3. Estimation of boundary layer transition noise from velocity measurements. Michael H. Krane and Wayne R. Pauley (Dept. of Aerospace Eng., Penn State Univ., 233 Hammond Bldg., University Park, PA 16802)

Velocity measurements of artificially generated flow structures in the transition region of an incompressible boundary layer with zero pressure gradient are described. These measurements made in a laminar flow water channel allow calculation of the velocity normal to the wall in a turbulent spot. This velocity specifies the linearized boundary condition for the acoustic equation at the wall. The approach relates the radiated noise to fluctuations in the normal velocity at the plate through fluctuations in the displacement thickness. Although this approach has been previously proposed [H. W. Liepmann, unpublished (1954), J. Laufer, J. E. Ffowcs-Williams, and S. Childress, *AGARDograph* 90, 39–42 (1964), G. C. Lauchle, *J. Acoust. Soc. Am.* 69, 665–671 (1981), G. C. Lauchle, *ASME NCA* 5, 31–38 (1989)] it has never been applied. The results of these experiments will be compared to concurrent experiments run in an anechoic wind tunnel. Ultimately this work will be extended to naturally occurring structures in the transition region. [Work supported by ONR under Grant #N00014-90-J-1365.]

9:35

6EA4. Radiation control of sound due to evolving instability waves along a curved surface. L. Maestrello (NASA Langley Res. Ctr., M.S. 463, Hampton, VA 23665-5225) and N. El-Hady (Analytical Services and Materials, Inc., Hampton, VA 23665-5225)

Active control of the acoustic pressure in the far field resulting from the growth and decay of a wave packet convecting in a boundary layer over a concave-convex surface is investigated numerically using direct computations of the Navier-Stokes equations. The resulting sound radiation is computed using the linearized Euler equations with the pressure from the Navier-Stokes solution as a time-dependent boundary condition. The acoustic far field exhibits directivity type of behavior that points upstream to the flow direction. A fixed control algorithm is used where the attenuation signal is synthesized by a filter which actively adapts itself to the amplitude-time response of the outgoing acoustic wave. This concept of active control may be applicable to wind tunnel contraction where noise radiated from the concave-convex surface is known to exist.

6EA5. Full-duplex speakerphone with acoustic and line echo cancelers. Sen M. Kuo and Jier Chen (Dept. of Elect. Eng., Northern Illinois Univ., Dekalb, IL 60115)

In this paper, a complete speakerphone is implemented with an AT&T WE DSP16A microprocessor. This single chip implementation has the advantage of low cost and hence can be applied to the speakerphone installed in a small conference room or a hands-free cellular phone. A robust speech detector is developed to classify four different operation modes; a corresponding gain regulation algorithm is then applied to guarantee the stability of local acoustic and electronic loop. More specifically, it reduces signal levels on both channels when in idle mode, freezes acoustic (or line) echo canceler during the transmit (or receive) mode, and freezes both echo cancelers when double talk mode occurs. The performance of the system is evaluated by computer simulation and tested by real time experiment.

10:05

6EA6. Choked-jet edge-tone experiments. Alan Powell and Dan Lin (Dept. of Mech. Eng., Univ. of Houston, Houston, TX 77204-4792)

Some experimental results for a choked jet from a 4.88:δ *rectangular* orifice impinging on a long 8° (w)edge are presented. For nozzle-edge distance $h > \text{three cell lengths}$, $< 15\delta$ and in the pressure ratio range 2.09 to 3.31, violent edge-tone oscillations occur that may be above, approximately equal to and merge with, or below the frequency of the simultaneous *single* screech mode of the free jet. The pressure field (\approx evanescent waves) of the growing sinusoidal jet instability are clearly visible in Schlieren photographs, as is the associated periodic asymmetrical sound field. Preliminary estimates indicate consistency with the proposed feedback formula for the fundamental acoustic wavelength λ [Powell, *Acoustica* 3, 233-243 (1953)], $h'/\lambda = (N + p) \times (M_{\text{con}})/(1 + M_{\text{con}})$ with: instability convection Mach number $M_{\text{con}} \approx 0.5-0.7$; nozzle to acoustic source distance $h' \approx h + \frac{1}{4} \times (\text{sinuous instability wavelength})$; integral number of cycles in the feedback loop $N = 4, 5, \text{ or } 6$ with $p \approx 0.4 \pm 0.2$. Normal hysteretic jumps appear negligible, but other hysteretic effects are suspected. [Partial support from the Texas Advanced Research Program.]

10:20

6EA7. Screech: Edge-tone-like behavior induced by a small plate. Alan Powell (Dept. of Mech. Eng., Univ. of Houston, Houston, TX 77204-4792), Y. Umeda, and R. Ishii (Dept. of Aeronautical Eng., Kyoto Univ., Kyoto 606, Japan)

The dominant screech frequency of *round* choked jets falls steadily with increasing pressure ratio, interrupted by jumps as the jet instability mode— A_1 and A_2 (toroidal), B (sinuous), C (helical), and D (sinuous again $\approx B$)—changes. A small normal plate (diam/nozzle diam = 0.35) on the axis radically changes the behavior. Preliminary results indicate that oscillations in the screech frequency range take place *only* in *four* “preferred” frequency bands, three being narrow of constant frequency, separated by prominent “dead” bands of *no* activity. With variable nozzle-plate distance, the *composite* plot for all pressure ratios shows *each composite band* to be reminiscent of classical sawtooth edge tones, with the number of cycles N in the apparent feedback loops from 2 to at least 8, with continuity of each N from band to band. But each band may contain *more than one screech-like mode*: Just toroidal for the band of highest frequency, then sinuous *and* helical, then sinuous, helical *and* sinuous again and finally just sinuous for the lowest frequency band. [Partial support from the Texas Advanced Research Program.]

6EA8. Reflection coefficient of a fluid-loaded anisotropic plate. D. E. Chimenti (Dept. of Mater. Sci. and Eng., Johns Hopkins Univ., Baltimore, MD 21218) and S. I. Rokhlin (Ohio State Univ., Columbus, OH 43210)

The acoustic reflection and guided wave behavior of an orthotropic plate immersed in an ideal fluid is examined as a function of fluid loading. Coincidence of zeros and real parts of reflection coefficient poles occurs only in the limit of vanishing or infinite fluid density. The modification of the pole and zero plate spectra in the presence of a variable fluid density will be discussed in the context of Lamb wave measurements on low density solid plates, such as graphite-reinforced plastics (GRP). Numerical results are presented showing functional dependence of pole and zero branches on fluid density and plate elastic properties at constant phase velocity or constant frequency-thickness product. It will be shown that propagating and evanescent wave modes exchange portions of their branches, and that reflection coefficient zeros also undergo extensive, but differing, transformations. Even very low fluid density ($< 0.1 \text{ g/cm}^3$) values can cause significant changes in the wave behavior of the fluid-loaded GRP plate. Fluid loading transforms the wave spectrum from Lamb waves in a traction-free plate to “constrained-slip” waves in a plate satisfying mixed boundary conditions.

10:50

6EA9. Measurement of unsteady pressure and shear stress using surface acoustic wave devices. Michael Macrorie, Vasundara V. Varadan, and Wayne R. Pauley (Dept. of Aerospace Eng., 233 Hammond Bldg., Penn State Univ., University Park, PA 16802)

A new technique is being investigated for the direct measurement of surface stresses (both pressure and shear) in steady and unsteady wall bounded flows. The technique utilizes surface acoustic wave (SAW) devices. A first generation sensor has been used to measure mean wall shear stress beneath a turbulent boundary layer at speeds from 10 to 20 m/s in air. Good agreement with a standard technique was found. To demonstrate the dynamic response, an artificially generated vortex structure was propagated over the SAW sensor. The time dependent pressure and shear stress oscillations due to the passage of the unsteady vortex were measured by the SAW sensor and compared with the response of a hot wire probe. Further reduction in the size of the sensors, multiple sensing elements, and refinement of the calibration technique will produce sensors capable of resolving the full stress vector due to near wall flow structures in transitional and turbulent boundary layers. [Work supported by ONR under Grant No. N00014-89-J-3102.]

11:05

6EA10. Scatter of subsonic waves on a fluid-loaded cylindrical shell from an internal obstacle. P. W. Smith, Jr. (Bolt Beranek and Newman Inc., 10 Moulton St., Cambridge, MA 02138)

An axisymmetric subsonic wave on a thin, fluid-loaded cylindrical shell of infinite extent impinges on a “shear diaphragm,” which constrains the normal velocity to vanish locally by the action of a ring force. The energy scatter coefficients (wave transmission, reflection, and sound radiation) have been evaluated, as well as the directional radiation source strength for a ring-force source. Results are presented for parameters appropriate to steel and water in the frequency range around the *in-vacuo* ring resonance frequency. The wave passes the obstacle nearly undisturbed at low frequencies ($ka < 1.5$, where a is the radius and k is the sound wave number). At high frequencies ($ka > 2.5$) the wave is largely reflected, behaving asymptotically like a flexural wave on a flat plate of the same thickness as the shell, with fluid on one side.

6EA11. Reflection, refraction, and transmission of vibrations by T junctions in piecewise-continuous beam constructions. B. C. H. Wendlandt (Dept. of Defence, Mater. Res. Lab., DSTO-Melbourne, P.O. Box 50, Ascot Vale, Victoria 3032, Australia)

A numerical analog to the laws of conservation and momentum and the constitutive laws of piecewise-continuous materials is developed and shown to be able to describe the propagation of vibrational waves in a composite. The analog considers material strains of up to 30% to permit a first-order description of nonlinear plastic and viscoelastic response of metal syntactic foam composite T junctions to relatively large scale vibrational and acoustic excitations. The computations describe the propagation of direct stresses across the junction and the generation of shear stresses at the T junction corners. Results of propagation studies of Gaussian and sinusoidal wave packets across a T junction are discussed. Results are presented for selected T junction composites. The results of the numerical analog are discussed in terms of an extension of the classical dilatation-torsion wave theory extended to piecewise continuous materials. Techniques to minimize reflections of vibrations or acoustic excitations from T junctions are discussed.

6EA12. Slowly varying method for high-frequency scalar scattering problems. Han Zhu (Dept. of Eng. Sci. and Mech., Univ. of Tenn., Knoxville, TN 37996-2030)

A new formulation called the slowly varying method (SVM) is proposed to asymptotically solve high-frequency scalar scattering problems. It is known that, excited by a plane harmonic incident wave $U_0(\mathbf{x})$ impinging on a scatterer, the scattered field $U(\mathbf{x})$ for Neumann B.C. or $\partial U(\mathbf{x})/\partial n$ for Dirichlet B.C. shows strong oscillation on scatterer when $ka \gg 1$. Here k is wave number and a is typical scatterer dimension. The paper shows that the function $V(\mathbf{x}) = U_0^*(\mathbf{x})U(\mathbf{x})$ for Neumann B.C. or $V_n(\mathbf{x}) = U_0^*(\mathbf{x})\partial U(\mathbf{x})/\partial n$ for Dirichlet B.C. exhibits much slow variation on scatterer for the same ka for some cases. Here $U_0^*(\mathbf{x})$ is the complex conjugate of $U_0(\mathbf{x})$. The relations among $V(\mathbf{x})$, $V_n(\mathbf{x})$ and the acoustic energy densities have been analyzed. A new boundary integral equation in terms of unknown $V(\mathbf{x})$ has been derived and either solved by BEM or simplified by asymptotical approximations. For the former case, the requirement that the boundary element dimension be proportional to $1/(ka)$ may be waived for obtaining satisfactory numerical results. For the latter case, a nonelement numerical solution of $V(\mathbf{x})$ has been obtained by an algebraic arithmetic. Some numerical results have been presented in the paper.

THURSDAY MORNING, 2 MAY 1991

INTERNATIONAL A, 9:00 TO 10:30 A.M.

Session 6ED

Education in Acoustics: Undergraduate Projects in Acoustics (Poster Session)

Mark F. Hamilton, Chair

Department of Mechanical Engineering, University of Texas, Austin, Texas 78759

All posters will be on display from 9:00 to 10:30 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 to 9:45 a.m. and contributors of even-numbered papers will be at their posters from 9:45 to 10:30 a.m.

6ED1. Diagnostic study of vibrational signals from a drill press. P. C. Marks, S. H. Angermund, V. Arya, S. D. Patel, M. G. Prasad, and C. Chassapis (Dept. of Mech. Eng., Stevens Inst. of Technol., Hoboken, NJ 07030)

The objective of this study was to measure the vibration of a drill press in order to detect the onset of chatter and to evaluate the use of vibrational measurements in a control system design. Chatter is unwanted vibration caused by the interference of the flank face of the bit with the bottom of the drilled hole which can be brought upon either by excessive feedrate or excessive drilling speed. Chatter is a major source of tool wear, which serves to increase machine downtime, thereby reducing productivity. Data were acquired in order to determine the baseline amplitude at which chatter occurs. Measurements were taken at feedrates varying from 9.4 mm/min, to 94.7 mm/min and with varying drill bit diameters. Chatter was seen to occur at feedrates over 65.8 mm/min for a worn bit and at over 94.7 mm/min for a new bit with an aluminum workpiece in a frequency range between 1400 and 2000 Hz. In a wood workpiece, no chatter was observed throughout the entire

range of feedrates. This study would be useful in the development of a feedback control system which would provide for drilling at the optimum feedrate for a given work piece based on chatter detection.

6ED2. Prototype and feasibility study of a PVDF infant health monitor. Joe Higgins and E. Carr Everbach (Dept. of Eng., Swarthmore College, Swarthmore, PA 19081-1397)

Piezoelectric polymer sheets, placed on the floor of a crib, can produce an output voltage that provides information about the heart and breathing rates of an infant in the crib. Such a device could be useful in detecting Sudden Infant Death Syndrome or other infant health problems. A prototype device has been developed with conditioning and detection electronics to isolate the contributions of each signal, and a feasibility study was developed to indicate whether such a device could be made cheaply and reliably enough to be commercially available to a wide population of consumers.

6ED3. Wave propagation in solids: Computation versus experiment. Ali Usman and E. Carr Everbach (Dept. of Eng., Swarthmore College, Swarthmore, PA 19081-1397)

Via two complementary methods, compressive and shear wave propagation in solids possessing complex geometries have been investigated. First, the finite-element package ANSYS was adapted to determine the stress field as a function of time due to a known excitation in solids with specified material properties. Then solids were constructed with the same properties and the strain field was measured as a function of time using fast strain gauges and laser interferometry. The results of these computational and experimental determinations will be compared.

6ED4. A computer controlled apparatus for measuring the angular scattering of ultrasound from a random medium. David R. Leroux and Murray S. Korman (Dept. of Phys., U.S. Naval Acad., Annapolis, MD 21402)

Measurements of the scattered sound intensity, $I(\theta)$, versus angle θ from submerged targets (such as a bubble cloud) require long tedious data runs. The task of generating a family of intensity curves (allowing for the adjustable parameters of a target) becomes insurmountable. An Apple IIe computer with an IEEE-488 instrument bus is programmed to control a digital oscilloscope (for data acquisition) and a stepper motor rotational unit (for angular positioning of the transmitting transducer). Typically, an average of 32 trials is used to measure the inco-

herent scattered energy from the received pulses at any particular angle. Scattering results from a cylindrical air bubble column are presented along with some theoretical predictions. [Work supported by the Naval Acad. Res. Council.]

6ED5. Scattering of sound by bubble clouds. Karin M. Hogan, Katherine L. Banta, and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD 21402)

Experimental results are shown for the scattering of ultrasonic pulses by a small volume of air bubbles in water. Two different types of bubblemakers are used to generate the bubble clouds. The first one uses an array of short wires spaced uniformly over a horizontal square area. A transient electronic pulse (of width τ) which has a period T , is simultaneously sent to each individual wire to generate microbubbles through electrolysis. The individual bubble radius generated at each wire is found to depend on the period of the repeated transient pulse. This bubblemaker is used to generate a vertical column of bubbles. The second bubble maker involves the regulation of a transiting volume of air stored in a chamber placed between two solenoid valves. The air escapes through a fine pore fritted disk to produce a transient bubble cloud. Preliminary scattering measurements are presented. Comparisons with theory are made from estimates of the void fraction β , the average bubble radius r_0 , and other geometrical parameters of the cloud. [Work supported by the Naval Acad. Res. Council.]

THURSDAY MORNING, 2 MAY 1991

INTERNATIONAL E, 8:00 TO 11:20 A.M.

Session 6NS

Noise: Industrial and Machinery Noise Control

Joseph Pope, Chair
P.O. Box 236, Newton Centre, Massachusetts 02159

Chair's Introduction—8:00

Invited Papers

8:05

6NS1. Noise control in the 1990's: A consultant's view. Lewis S. Goodfriend (Lewis S. Goodfriend & Assoc., P.O. Box 2453, Morristown, NJ 07962-2453)

The acoustical consultant working with both manufacturers and users of industrial machinery must become thoroughly conversant with the client's equipment or facility and then, and only then, can he take the responsibility for recommending process changes and equipment modifications that will result in noise control. Notwithstanding advances in the analytical and instrumentation areas related to noise propagation and transmission, the methods of noise control other than source modification are, in general, similar to, or the same, as those used in the two preceding decades. Some new hardware has become available including noise cancellation equipment. However, no new theories of noise control have appeared, and the materials and hardware available for noise control have changed little over this period. Some proprietary flow control technology is just now appearing. In general, however, improvements in older materials and systems has simplified the use of existing technology. Industries in the chemical process and utility field still depend on passive "silencers" and insulating jacketing to achieve noise control. Active noise control methods are still mostly confined to experimental work. The use of modern instrumentation has simplified the identification of internal equipment sources. The characteristics of today's methods, materials, and equipment are reviewed along with the expected results. It remains the consultant's responsibility to integrate the noise control measures into the equipment, system, or process without adverse effects on the operation. Several examples from industrial applications will be described.

6NS2. Progress in noise control of mining equipment. Leonard C. Marraccini (Mine Safety and Health Admin., Pittsburgh Health Technol. Ctr., 4800 Forbes Ave., Pittsburgh, PA 15213)

With the advent of mechanization in the mining industry, one of the by-products has been the generation of excessive noise levels. In order to protect miners from these excessive noise levels, the Mine Safety and Health Administration has promulgated various noise regulations over the years, starting with the Coal Mine Safety and Health Act of 1969. One method used to reduce these excessive noise levels has been with the utilization of engineering noise controls. The progress in noise control of mining equipment has essentially followed three avenues of approach. These involve retrofit noise control, machine redesign, and advanced technology. One example of each of these approaches is (1) cut and fit application of acoustical materials to mobile equipment cabs, (2) redesign of cutting heads for underground continuous miners, and (3) electronic noise cancellation. This presentation will involve a general discussion of the progress made in noise control of mining equipment. In addition to the three examples previously mentioned, other examples of noise control will also be discussed.

6NS3. Mathematical analysis of machinery noise. A. F. Seybert (Dept. of Mech. Eng., Univ. of Kentucky, Lexington, KY 40506-0046)

As an aid to understanding and controlling machinery noise, mathematical methods may be used. Because of the complex geometry involved, the methods are normally implemented numerically, with solutions obtained on the computer. Two such methods are the finite element method and the boundary element method. This presentation will review and compare these two methods as they are generally applied to machinery noise prediction. The development and practical application of the methods during the past five years will be discussed. Examples illustrating where each method has advantages or disadvantages will be presented. The role of mathematical and computer analysis of machinery noise in the future will be discussed, including the limitations imposed by lack of easy-to-use software and fast computers.

6NS4. Simulation of the radiated noise emitted by two coupled structures and application to the acoustic quietening of air conditioning equipment. J. Nicolas (GAUS, Département de génie mécanique, Université de Sherbrooke, Sherbrooke, Québec J1K 2R1, Canada)

The challenge of noise control for the nineties will be to leave trial and error methods that are efficient in specific well-known situations and to tackle the difficult and complex but inevitable problem of structural acoustics and vibration. In order to actually control the noise at the source at the design step, a mode of the radiation of a semicomplex structure has been developed. Based on an analytical approach using a variational method, this model allows prediction of the effects of the boundary conditions [J. Acoust. Soc. Am. (to be published)] as well as stiffeners and added masses with force or moment type of excitation. To go toward industrial applications, the calculus has been extended to the case of a mechanical source of vibration (electric motor, engine) which is attached to a large thin structure that acts as a noise radiator. A theoretical analysis of the problem is presented. The actual force input into the structure is determined as the resultant of both the output impedance of the source and the input impedance of the structure. A quadripole approach for the source assembly enables calculation of the force input, the kinetic energy, the radiation efficiency, and overall sound power. The key novelty of this method lies in its capacity of predicting *a priori* the radiated noise in various configurations, allowing the designer to choose rationally the best configuration in terms of noise control. This approach has been applied in order to decrease the unwanted noise emitted by air conditioning equipment. A new design has been proposed which led to a 10-dB improvement. In a general case, guidelines for the optimal suspension design will be presented.

6NS5. History, progress, and development of a standard for measurement of the noise of new machinery at the operator's position. Stephen I. Roth (Aluminum Co. of America, 1501 Alcoa Bldg., Pittsburgh, PA 15219)

An engineer should be able to request reliable and accurate equipment noise level data from a manufacturer to determine equipment acceptance. If analysis of the data indicates excessive noise, the engineer should proceed with an approach to insure an acceptably quiet installation. ANSI Working Group S12/WG20 is writing a standard to assist an engineer in obtaining noise level data of new, stationary equipment. ANSI S12.20 will incorporate the use of existing ANSI, trade association, and professional society noise measurement standards. It will incorporate guidelines to allow the user to interpret equipment noise level data. It is hoped that if users frequently request noise level data using ANSI S12.20, manufacturers will be better prepared to provide noise level data resulting in more accurate, consistent, and timely noise level information for user evaluation.

6NS6. Quiet zones using active control. Colin F. Ross (Noise Cancellation Technologies Inc., 1015 W. Nursery Rd., Linthicum, MD 21090)

This paper will review the recent advances in active control technology that allow small zones to be protected in a noisy environment. It will compare the technological approaches that are appropriate to the different types of noise (broad and narrow band) and the source of the noise (distributed or point). The paper will concentrate on the fundamental limits imposed by causality and signal to noise on the performance of such systems. It will finally compare the typical performances available from the different types of system.

Contributed Papers

10:35

6NS7. A cavity-piston model for acoustical source impedance in a duct. M. G. Prasad and K. Wu (Dept. of Mech. Eng., Stevens Inst. of Tech., Castle Point on the Hudson, Hoboken, NJ 07030)

The importance of acoustical source impedance in the performance evaluation of a muffler in terms of insertion loss is well known. Several experimental methods have been used to obtain the acoustical source impedance. However, studies based on analytical approaches are not many. This paper presents a cavity-piston impedance model for an acoustical source based on the combined pressure-velocity source behavior. The present study deals with the analysis of a source comprising a finite length pipe with an oscillating piston as the termination. The impedance curves for the case of oscillating piston reduces in the limit to the well known case of pipe with rigid end termination. The parameters studied are frequency and amplitude of the piston. The application of this study to noise sources are discussed.

10:50

6NS8. A coherent structure model to explain the quadrupole directivity of noise from impinging jet. Jianping Shen, Seungbae Lee, and William C. Meecham (Dept. of Mech., Aerospace, and Nucl. Eng., Univ. of Cal., Los Angeles, CA 90024)

The quadrupole-like directivity patterns of jet noise, created by jet impingement on a large flat plate were experimentally obtained by the authors previously [J. Acoust. Soc. Am. Suppl. 1 88, S8 (1990)]. Theoretical considerations have been examined to explain physically this directivity phenomena, and a model coherent structure called the "orientated vortex motion" is proposed. The model hypothesizes that a train of partially organized, large-scale vortices exists inside the boundary layer flow near the plate. By including their images in the rigid plate

an oscillation is formed, resulting in a quadrupole acoustical source, which explains the experiments quite well. This is analogous to the low Reynolds number free jet case that predicts a possible, similar directivity pattern for the noise from the free jet. A large eddy simulation (LES) computational technique for the simulation of turbulent flows seems to be a feasible approach to investigate the proposed organized motion. A simulated statistically stationary flow field will be discussed.

11:05

6NS9. Sound level measurements in background noise. Michael A. Staiano (Staiano Engineering, Inc., 1923 Stanley Ave., Rockville, MD 20851)

In many sound measurement situations, a need arises to determine the magnitude of a specific sound source while persistent—although fluctuating—background sound exists due to other sound sources. Usually the specific source can be turned off but no control of the background noise is possible. As a result, measurements can be obtained of the "total" (consisting of the source and background together) and of the background alone. Although "survey methods" for the determination of sound power levels permit background noise "at least 3 dB below the A-weighted sound pressure level with the source operating," even this requirement may be too stringent for some situations where estimates of band sound levels are required. Mathematically a correction can be derived in any situation where a measurable difference exists between the as-measured total (that is, the source + background) and the background noise levels. Where the difference between the total and the background noise is small, the accuracy is poor. However, if a confidence interval (i.e., a range in which the correct result is likely to occur) can be defined, the estimated result may be of use. This paper explores a scheme for estimating and reporting source sound levels measured in the presence of background noise.

Session 6PAa

Physical Acoustics: Scattering I

Michael R. Stinson, Chair

National Institute for Microstructural Sciences, National Research Council, Ottawa, Ontario K1A 0R6, Canada

Contributed Papers

8:30

6PAa1. Ray synthesis of backscattering by thin cylindrical shells. N. H. Sun and P. L. Marston (Dept. of Physics, Washington State Univ., Pullman, WA 99164-2814)

In previous research [P. L. Marston, *J. Acoust. Soc. Am.* **83**, 25–37 (1988)] the leaky Lamb wave contributions f_l to the form function for backscattering from right circular cylindrical shells were expressed in terms of the phase velocity ratio c_l/c , a radiation damping parameter β_l in Np/rad, and a coupling coefficient G_l with $|G_l| \approx 8\pi\beta_l/(\pi ka)^{1/2}$. It was subsequently shown that the phase $\arg(G_l) \approx \pi/4$. The present research concerns the extension to contributions termed “creeping and trapped” waves from the far-field limit of an analysis (when corrected) by Felsen *et al.* [*J. Acoust. Soc. Am.* **87**, 554–569 (1990)]. Such contributions are especially important for understanding scattering from thin shells. The extension of Marston’s original equations requires that when $c_l < c$, the effective angle of incidence θ_l for launching a guided wave becomes $\pi/2$ instead of the previous expression $\sin \theta_l = c/c_l$; also, $\arg(G_l)$ is revised when $c_l \leq c$. For example, for trapped waves $\arg(G_l)$ becomes $\approx -\pi/4$. The expression for f_l allows the caustic radius $b_l = ac/c_l$ to be larger than the radius a of the shell. Comparisons with exact computations for 2.5% thick hollow steel cylinders support the approximations for the leaky and trapped wave contributions. The ka dependence of leaky, creeping, and trapped wave parameters are examined. [Work supported by ONR.]

8:45

6PAa2. Ray synthesis of the angular dependence and backscattering by thick cylindrical shells. P. L. Marston and N. H. Sun (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

Previous research has confirmed ray models of leaky Lamb wave contributions f_l are applicable to the forward and backward scattering from thick spherical shells [S. G. Kargl and P. L. Marston, *J. Acoust. Soc. Am.* **88**, 1103–1113 (1990); and work in press]. The present research concerns the extension of the ray models to leaky Lamb waves that tend to dominate the elastic scattering contributions of thick cylindrical shells. The extension is important because the cylinder model should be generalizable to noncircular-shapes where solutions for the scattering may not be available. A previous analysis found that for cylinders, the coupling coefficient becomes $|G_l| \approx 8\pi\beta_l/(\pi ka)^{1/2}$ with the phase $\arg(G_l) \approx \pi/4$, where β_l is the radiation damping parameter. The present computations show that for a 16.2% thick stainless-steel cylinder, a superposition of leaky and specular contributions synthesizes the exactly computed scattering for ka as small as 7. The scattering as a function of angle was synthesized at $ka = 20$ for the full 180° range. The synthesis combined leaky Lamb wave contributions with a background contribution of a rigid cylinder. The agreement with exact com-

putations was good except near ray transition regions where paths changed abruptly with changes in the scattering angle. [Work supported by ONR.]

9:00

6PAa3. Scattering by thick cylindrical shells: Ray methods for shell reverberations and experiments. P. L. Marston, N. H. Sun, S. P. Parry, and D. H. Hughes (Dept. of Physics, Washington State Univ., Pullman, WA 99164-2814)

A previous analysis of internal reverberations [S. G. Kargl and P. L. Marston, *J. Acoust. Soc. Am.* **88**, 1114–1122 (1990)] gave the curvature correction to the specular contribution f_{sp} to the form function for backscattering from spherical shells. This was extended to the case of circular cylinders at normal incidence. The result for f_{sp} is $R \exp(-2ix) + f_{cc}$ where the curvature correction is

$$f_{cc} = \left(\frac{re^{in}}{1 - re^{in}} - \sum_n \frac{r^n e^{in\alpha}}{(1 + nB)^{1/2}} \right) \left(\frac{1 - r^2}{r} \right) e^{-2ix},$$

with $x = ka$ and the other symbols are as defined for Eq. (8) of the previous analysis for fluid spherical shells. Comparisons with exact partial-wave series computations for fluid shells show good agreement. They exhibit a dip in $|f_{sp}|$ when x is small and when x approaches the thickness resonance. The features are generally similar to those for spheres. The relevance of f_{sp} to the calibration of tank experiments on the scattering of short tone bursts from elastic shells is examined. The experiments separate leaky Lamb wave contributions from specular contributions. [Work supported by ONR.]

9:15

6PAa4. Analysis of the acoustic scattering from large aspect ratio, axisymmetric shells. Roger H. Hackman and T. Fang (Palo Alto Res. Labs., Orgn. 91-50, Bldg. 251, 3251 Hanover St., Palo Alto, CA 94304-1191)

The development of theoretically sound, “exact” numerical solutions to the scattering from spheroidal shells is important both to establish a nontrivial benchmark and to establish those salient features that may be characteristic of a larger class of more complicated, but related geometries. In previous meetings [*J. Acoust. Soc. Am. Suppl.* **1** **83**, S94 (1988); **82**, S49 (1987)], the predictions of the spheroidal-coordinate based transition matrix formalism was validated for such targets by a direct comparison with experimental measurements of the backscattered form function of a copper prolate spheroidal shell for general angle of incidence. In these studies, a number of open questions emerged pertaining to the prominent elastic features of the form func-

tion of the shell. These questions are revisited here and the elastic response of the shell in terms of the modes of a fluid-loaded, infinite cylindrical shell is analyzed. In this work, it is particularly useful to study the evolution of the scattering as the thick shell limit is approached with reference to the scattering solution (and modes) of a solid elastic target. By examining the evolution of the prominent features of the scattering solution, the evolution of the underlying modes of the system can be inferred. This evolution shows a strong analogy to the evolution of the modes of a spherical shell presented previously.

9:30

6PAa5. Abstract withdrawn.

9:45

6PAa6. Differential scattering cross sections at arbitrary outgoing angles and their relation to target boundary conditions. Jacob George and M. F. Werby (NOARL, Code 221, Stennis Space Ctr., MS 39529)

It is usual to examine differential scattering cross sections in the backward direction as a function of frequency variation. When this is done in the asymptotic limit the resulting function is referred to as a form function. The form function proves to be sensitive to the boundary conditions of the target and may manifest resonances and circumferential diffraction effects typical of certain targets. However, such quantities at exit angles such as those in the forward direction prove to be somewhat less sensitive to the object's properties with increasing frequency. This characteristic is undesirable if one wishes to examine the material nature of the target. On the other hand, it can be valuable from the modeling point of view if one wishes to obtain forward scattering target strengths. In that case one may use the simplest target to obtain such a property. Forward and other exit angle target scattering strengths were examined over a frequency region to determine when the strengths begin to converge and thus when the simple approximations can be made.

10:00

6PAa7. Determination of material composition from time-domain and frequency-domain resonance echoes of submerged elongated elastic targets. C. E. Dean and M. F. Werby (NOARL, Code 221, Stennis Space Ctr., MS 39529)

When scattering from elastic targets backscattered echoes yield interesting information in the resonance region. In particular, resonance scattering theory in the frequency domain along with the circumferential nature of resonances imply that material constituency is a charac-

teristic of resonance location. Moreover, recent discussions of resonance signatures in the time domain [see Uberall's book to be published on resonance scattering] can also yield information concerning resonance widths and average phase velocities which can also be related to material characteristics. By adjusting the orientation of the target over a suitable angular region it is possible to ascertain certain symmetries of the target if they exist, particularly if one varies the frequencies over a suitable range of resonances. If one observes axial symmetry through such a process, then it is possible to obtain both the dimensions of the object and the aspect ratio of the object (ratio of length to width). This is assuming that the target is in a "free" environment; that is, the boundaries of the target are not a factor in calculation. Time-domain responses for specific pulse types also yield information and it is easy to see how a series of questions can form the basis of a scenario that can rule out certain targets or lead to a probability (confidence level) that specific targets are present. To determine the extent that this can be done, targets are examined which are composed of five materials for elastic solid spheroids for aspect ratios of 3 to 1 and 6 to 1 for end-on incidence and for the case of 4 to 1 for all incident angles.

10:15

6PAa8. Constrained total least-squares-based extraction of acoustic scattering resonances of elastic targets. Theagenis J. Abatzoglou and Roger H. Hackman (Palo Alto Res. Labs., Orgn. 91-50, Bldg. 251, 3251 Hanover St., Palo Alto, CA 94304-1191)

The analysis of the resonance response of elastic targets to an insonifying beam has been a central theme in acoustic scattering investigations in recent years. The motivation for this work is that the resonance structure of such targets is directly related to both the geometry and material properties of the target and this has a bearing on target classification. The development of robust techniques to estimate the resonance frequencies from the transient scattered field in the presence of noise is an important and open problem that is the subject of the current study. In the following, the efficacy of several competing techniques are examined (i.e., the constrained total least squares [T. J. Abatzoglou *et al.*, IEEE Trans. ASSP (May 1991)], the Prony, and a recently developed singular value decomposition based technique), with theoretically calculated scattering data. To simulate scattering data, the spheroidal coordinate based *T*-matrix formalism is utilized [R. H. Hackman and D. G. Todoroff, J. Acoust. Soc. Am. 78, 1058-1071 (1985)] to compute the impulse response of large aspect ratio, solid elastic cylinders for target aspects ranging from end-on to broadside; additive Gaussian noise is included in the time domain representation. Such scatterers pose realistic challenges in that their target signature varies strongly with frequency and target aspect. Both monostatic and bistatic scattering are considered.

10:30

6PAa9. Determination of the vibration mode by a short pulse method. P. Rembert, P. Pareige, G. Maze, and J. Ripoche (Lab. d'Acoust. Ultrasonore et d'Electron., URA CNRS 1373, Université du Havre, Place R. Schuman, 76610 La Havre, France)

It is well known that several experimental methods using quasi-harmonic insonification allow a direct verification of the resonance scattering theory. They all provide, for elastic cylindrical targets immersed in water, the resonance frequencies and the mode number of each one. So far, pulsed techniques only allowed resonance isolation. Presented here is a new method: the short pulse method of isolation and identification that completes these works and allows a total comparison with theoretical and quasi-harmonic experimental results. This method consists in the digitizing of the time signals that characterize the scattering of an elastic target insonified by a short pulse for different angular positions of the receiving transducer. For each so obtained signal, a resonance spectrum is obtained after a spectral amplitude analysis. From these data, a computer processing allows the plotting of identification patterns, and, by the same way, the knowledge of the mode of vibration at a given resonance frequency. This method has been used for the identification of

the resonances of targets as cylinders, water-filled or air-filled cylindrical shells, cavities, etc. The results obtained by this new short pulsed method of identification agree with all those already published.

10:45

6PAa10. Intelligent ultrasonic signal processing for solving inverse scattering problems. Wolfgang Sachse (Dept. of Theoret. and Appl. Mech., Cornell Univ., Ithaca, NY 14853) and Igor Grabec (Univ. v Ljubljani, Ljubljani, Yugoslavia)

This paper describes new *intelligent* or *neural-like* processing procedures by which the ultrasonic signals scattered by an obstacle can be processed to recover the characteristics of the unknown scatterer. In particular, it focuses on the recently developed *automatic modeler* by which the characteristics of the scattering phenomenon from a number of scatterers is learned using a set of systematic learning data. It is demonstrated that ultrasonic signals measured in subsequent tests from an unknown scatterer can be processed to optimally recover the characteristics of the unknown obstacle. The solution of the inverse scattering problem that is obtained is thus completely empirical and not based on any elastodynamic theory. [Work supported by the National Science Foundation under Grant MSM-8904384.]

11:00

6PAa11. Characterization of white blood cells using inverse acoustic scattering. Xucai Chen, Robert E. Apfel (Dept. of Mech. Eng., Yale Univ., P.O. Box 2159, New Haven, CT 06520), and Stephen Wardlaw (School of Medicine, Yale Univ., New Haven, CT 06510)

The acoustic scattering functions of human leukocytes (white blood cells) in isotonic saline solution are detected at two angles, along with information on their volume obtained from electrozone sensing. The experimental apparatus has been reported earlier [J. Acoust. Soc. Am. Suppl. 1 **84**, S163 (1988)]. Different procedures are used to invert the experimental data. Multidimensional histograms are generated and patterns in these histograms are used to recognize white cell subgroups. This technique may be developed into a differential hematology analyzer or combined with currently available analyzers to improve the measurement confidence. [Work supported by the U.S. National Institutes of Health through Grant 5R01CA39374.]

11:15

6PAa12. Optimization of the distance between transducer and scatterer in hydroacoustical modeling. Vladimir Genis (Biomed. Eng. and Sci. Inst., Drexel Univ., 32nd and Chestnut St., Philadelphia, PA 19104), I. Oboznenko (Kiev Polytech. Univ.), J. M. Reid, R. Raghavan, K. Soetanto, H. Oung, and X. M. Lu (Drexel Univ., Philadelphia, PA 19104)

This work was motivated by the need to understand the effect of phase and amplitude variations on the scatterer's surface as well as the reflecting boundaries of the hydroacoustical tank on the structure of the acoustical field in modeling. It is known that the null positions of the spectral characteristics of the backscattered signals from various objects in water can yield information about their shape, dimensions, orientation, and structure. Accuracy of such measurements is affected by the amplitude and phase variations on the scatterer surface and these variations depend on the minimum distance between the transmitter and scatterer positions. Calculations of these distances for acceptable dimensions of acoustical tanks were performed. Expressions were derived to correct the minimum allowable distance to prevent time coincidence of impulses from the scatterer and from the walls of the echoic acoustical tank.

11:30

6PAa13. Making complex underwater scattering targets out of polystyrene foam. H. R. Suiter (Naval Coastal Syst. Ctr., Code 2120, Panama City, FL 32407)

A method is described of casting polystyrene beads in small quantities for laboratory use as scattering targets. The method is easy and requires no specialized equipment other than a mold. Two such targets are described. One of them is a 4-period shallow wedge grating and the other a 7-period phase grating constructed with acoustically hard aluminum strips imbedded in the acoustically soft foam. Results of a tank experiment on these highly reflective targets will be given. The measured backscattered reflectivities are compared with simple Huygens-Fresnel-Kirchhoff diffraction theory in the high ka regime where the Kirchhoff approximation should work well. Agreement is satisfactory for the wedge gratings. For the phase grating, the facet width must be used as a parameter to reproduce the higher-order behavior. [Work supported by ONR.]

THURSDAY MORNING, 2 MAY 1991

PRATT A & B, 9:00 TO 11:30 A.M.

Session 6PAb

Physical Acoustics: Outdoor Sound Propagation

W. Pat Arnott, Chair

National Center for Physical Acoustics, Coliseum Drive, University, Mississippi 38677

Contributed Papers

9:00

6PAb1. One-dimensional model of turbulence-induced distortion of sonic boom profiles. Andrew A. Piacsek (Grad. Prog. in Acoust., Penn State Univ., P. O. Box 30, State College, PA 16804)

In a recent paper (AIAA-90-4031), Pierce and Sparrow have sug-

gested that the prediction of the distortion of sonic booms by atmospheric turbulence can be carried out using modification of the Friedlander series theory that gives waveforms as a power series in time relative to wave onset. The present paper explores basic mathematical problems associated with that theory by analysis of a simplified mathematical model. A step pulse in acoustic pressure propagating in the

+ x direction enters at time $t = 0$ the region $x > 0$ in which sound speed $c(x)$ is a random function of position, where this function is drawn from an ensemble with given statistical properties. The wave at any positive x begins with a weak shock that arrives at a time τ equal to the x integral of $1/c$ and which, for $t < \tau$ can be expanded in a power series $t > \tau$. The statistics of the coefficients in this power series are studied with the assumption that the random process $c(x)$ is homogeneous and Gaussian.

9:15

6PAb2. Molecular relaxation is insufficient to explain the shock structure and rise times of sonic booms; turbulence is apparently important. Thomas A. Gionfriddo, Jongmin Kang, Victor W. Sparrow, and Allan D. Pierce (Grad. Prog. in Acoust. and Dept. of Mech. Eng., Penn State Univ., P. O. Box 30, State College, PA 16804)

The authors have been recently examining some sonic-boom waveforms that were recorded during overflights by the Air Force and that have become available to NASA and its contractors. The quality of the digitized data and the supporting meteorological data was such that one could test the applicability of molecular relaxation theories. In the late sixties it had been supposed that the finite rise times of sonic booms was attributable to atmospheric turbulence, but it was later pointed out that the first estimates of rise times in the absence of turbulence had neglected the vibrational relaxation of nitrogen molecules. Bass *et al.* [J. Acoust. Soc. Am. 74, 1514–1517 (1983)] have demonstrated that molecular relaxation definitely gives the correct order of magnitude of the observed rise times. However, the Air Force data in conjunction with the recent steady-state shock profile model theory of Kang and Pierce give the first opportunity to make a detailed quantitative assessment of the molecular relaxation hypothesis. The agreement of theory with experiment in some cases is remarkably excellent, but in the preponderance of cases the rise of the shock is slower and the rise time is longer, typically by factors of the order of 2 to 3. [Work supported by NASA-LRC and by the William E. Leonhard endowment to Penn State Univ.]

9:30

6PAb3. A statistical study of the relation between weather conditions and sonic boom waveforms. Lixin Yao, Henry E. Bass, and Richard Raspet (Phys. Acoust. Res. Group, Dept. of Phys. and Astron., The Univ. of Miss., University, MS 38677)

Sonic boom data collected at Edwards Air Force Base in Oklahoma City in the 1960s have been examined to check the validity of some common assumptions concerning the role of turbulence on sonic boom waveforms. A Turner Class was assigned to each flyby based upon reported ground weather conditions. This Turner Class was then used to indicate the presence of convective or mechanical turbulence or stable stratification. The correlation between the Turner Class and the waveform and rise time was then calculated. These results indicate that mechanical turbulence is associated with sonic booms that have more rounded shapes and a greater rise time than common for stable conditions. [Work supported by NASA Langley Research Center.]

9:45

6PAb4. Fourier transforms and effective rise times of sonic booms propagating through a relaxing atmosphere. Jongmin Kang (Noise Control Lab., Dept. of Mech. Eng., Penn State Univ., 157 Hammond Bldg., University Park, PA 16802)

Sharp impulsive sounds, such as sonic booms, typically have sudden pressure jumps (shocks) that are principal contributors to their perceived annoyance. Rise time, the ostensible time over which the sudden pressure rise occurs, is somewhat of an inadequate descriptor because not all profiles are similar and because two profiles with the same overpressure and rise time may seem markedly different in perceived loud-

ness. A previously developed numerical procedure determines the detailed pressure versus time profile during the interval that the leading shock nominally occurs, taking into account molecular relaxation effects, and the Fourier transform is determined using the known asymptotic behavior of the waveform profile immediately preceding and following the shock. The proposed definition of an effective rise time recognizes that the A-weighted sound exposure of a waveform is a good descriptor of perceived loudness and that Fourier transforms of pressure waveforms with shocks with finite rise times tend to fall off as the inverse square of increasing frequency as one over the square of the frequency. Such considerations lead to a definition of rise time that is inversely proportional to the square root of the A-weighted sound exposure of that portion of the waveform which includes the shock. The basis for this definition is explained, the constant is evaluated, and examples of atmospheric sonic booms are discussed. [Work supported by NASA-LRC and by the William E. Leonhard endowment to Penn State Univ. The author acknowledges the advice of A. D. Pierce.]

10:00

6PAb5. Propagation in atmospheric convective-boundary-layer turbulence. D. Keith Wilson and Dennis W. Thomson (Dept. of Meteorology, Penn State Univ., 503 Walker Bldg., University Park, PA 16802)

Turbulence in the atmospheric convective boundary layer is inhomogeneous and anisotropic. The vertical correlation length of the sound speed field is typically proportional to height from the ground, whereas the horizontal correlation length is proportional to boundary-layer thickness. Solutions for the turbulence strength and diffraction parameters, valid for this particular case, are presented. The solutions differ significantly from those for homogeneous, isotropic turbulence. An intercomparison is made between the theoretical solutions, numerical propagation simulations, and experimental data. For the numerical simulations, rays were traced through wind and temperature fields generated by a large-eddy simulation.

10:15

6PAb6. Abstract withdrawn.

10:30

6PAb7. Analysis of wind noise measurements. Scott Morgan and Richard Raspet (Phys. Acoust. Res. Group, Dept. of Physics and Astron., The Univ. of Miss., University, MS 38677)

Wind noise in outdoor microphone measurements is composed of both the noise due to intrinsic turbulence in the flow and that due to the flow interacting with the microphone. This paper presents the analysis of wind noise and turbulence data taken in an outdoor environment in an effort to determine the primary source of the wind noise. Bare microphone data are compared with data taken during the same time period from microphones covered with a nose cone, a Brüel and Kjaer spherical windscreen, and a variety of experimental windscreens developed at the University of Mississippi. The theory behind these experimental screens is also presented and is compared to the experimental data. Conclusions concerning the dominant source of wind noise are drawn by examining microphone data and turbulence data and comparing them with published microphone data. The effectiveness of different windscreens is also examined. [Work supported by MIT Lincoln Laboratory.]

10:45

6PAb8. The influence of an ideal wind profile on sound propagation over a ridge. M. J. Jaye (Dept. of Math. Sci., USMA, West Point, NY 10996), M. J. Jacobson (R. P. I., Troy, NY 12180), J. S. Robertson (USMA, West Point, NY 10996), and W. L. Siegmann (R. P. I., Troy, NY 12180)

The effects on low-frequency acoustic propagation resulting from ideal atmospheric flow over a large ridge are investigated using the parabolic approximation. The ridge is taken to be triangularly shaped with a horizontal earth-air interface on both sides. A Schwarz-Christoffel transformation is employed to calculate the wind speeds that are then used to compute the effective sound-speed profiles. These profiles are used by an implicit finite-difference implementation of the parabolic approximation to estimate the intensity of the sound field. Several examples are examined to determine the effects of this wind-modeling

method on sound pressure levels over rigid earth-air boundaries. [Work supported by NASA.]

11:00

6PAb9. Low-frequency long-range sound propagation over impedance discontinuities with the parabolic approximation. P. J. Schlatter, J. S. Robertson (Dept. of Math. Sci., USMA, West Point, NY 10996), and W. L. Siegmann (R. P. I., Troy, NY 12180)

Previous studies have shown that the parabolic approximation method, widely used in ocean acoustics, can be successfully applied to low-frequency long-range atmospheric sound propagation over a locally reacting boundary, an important problem with many applications. This work models cw acoustic signals as they propagate over horizontal ground surfaces with impedance discontinuities. An implicit finite-difference implementation of the parabolic approximation sound propagation model incorporating locally reacting, variable-impedance ground surfaces is employed to compute estimates of the sound field excess attenuation for a variety of flow resistivities and source frequencies. Both windy and stationary atmospheres are considered. An accuracy benchmark and several example problems in which the sound propagation path includes an idealized lake will be discussed. [Work supported by NASA.]

11:15

6PAb10. Acoustics of marine fog: Present standards of transmission, reception, and projection. David G. Browning (Marine Sci. Inst., Univ. Conn., Groton, CT 06340) and Michelle Fitzpatrick (U.S. Coast Guard Acad., New London, CT 06320)

The standards for sound propagation in marine fog have been in service for many years. For example, computations of propagation loss are based on the work of Wiener [F. M. Wiener, J. Acoust. Soc. Am. 33, 1200-1205 (1961)]. Similarly, standards for reception and projection of acoustic signals in marine fog are based on work by Bolt Beranek and Newman Inc. in 1960. A review is made of results that has been published since that time to see if any possible improvements have been suggested.

Session 6SA

Structural Acoustics and Vibration: Modal, Wave-Vector, and Signal Processing Methods

Leopold B. Felsen, Cochair

Department of Electrical Engineering and Computer Science, Polytechnic University, Route 110, Farmingdale, New York 11735

Arthur B. Baggeroer, Cochair

Department of Ocean and Electrical Engineering, Massachusetts Institute of Technology, Cambridge, Massachusetts 02139

Chair's Introduction—8:30

Invited Papers

8:35

6SA1. Hybrid ray-mode resonance system format for the acoustic response of submerged structures. L. B. Felsen^{a)} (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

Many submerged structures of interest are comprised of elastic shells with attached loadings and/or substructures. In the midfrequency range, traveling waves in shell segments between loads, and also in substructures, may become important, whereas complex loadings as such may be better characterized in terms of their modal resonances. This motivates the formulation of a self-consistent hybrid ray-(traveling mode)-resonance system that highlights the physical wave phenomena responsible for synthesizing the interactive fluid-structure acoustic response. Such an observable-based parametrization (OBP) generalizes the conventional approach based on modal resonances alone and reorganizes resonances compactly into propagators where these predominate. The OBP format for cw and transient excitation is illustrated on the example of an aperture-coupled ring-waveguide cavity in a rigid cylindrical baffle developed originally for electromagnetic radar scattering where the method was first applied [L.B. Felsen and G. Vecchi, IEEE Trans. Antennas Propag. (in press)]. Especially in the time domain, where successive early interactions are resolved by arrival times whereas later interactions are reconstituted as resonances, the ability of OBP to identify the physical observables is clearly evident. Similar behavior characterizes sound scattering from a submerged spherical elastic shell. [Work supported by ONR.] ^{a)} Visiting.

9:00

6SA2. Modal and wave approaches to the analysis of complex structures. G. Maidanik and J. Dickey (DTRC, Bethesda, MD 20084)

A complex structure may be analyzed by decomposing the structure into coupled elemental substructures, and, even further, into coupled elemental dynamic systems. To the latter decomposition a modal and a wave method of analysis may be applied. The two methods are complementary and supplemental. The analogies and the differences between the modal and the wave approaches will be discussed.

9:25

6SA3. Frequency window analysis of submerged shells with internal subsystems. Jens Bjarnason, Takeru Igusa, and Jan D. Achenbach (Dept. of Civil Eng., Northwestern Univ., Evanston, IL 60208)

The far-field radiation from harmonically loaded submerged shells with internal subsystems is studied. The analysis is based on modal expansions derived from Lagrange's equations. The complexity of the problem is reduced by separating terms with respect to a frequency window. Detail is retained for terms within the window, and dominant coupling effects are included for terms outside the window. This frequency window method provides mathematical insight into two important characteristics of the fluid-shell-substructure system: the modal interaction between the substructure and the shell, and the transition across frequency scales from inefficient to efficient radiating shell modes. The study shows that subsystems have significant effects on the acoustics of submerged shell structures, and that the frequency window method is a useful modal approach for analyzing these effects. [Work supported by ONR.]

6SA4. Array processing for wave vectors and modes in structural acoustics. Arthur B. Baggeroer (Depts. of Ocean and Elec. Eng., MIT, Cambridge, MA 02139)

Array processing methods have emphasized beamforming and directional spectrum estimation for equally spaced, linear arrays with single component sensors that are in a homogeneous signal field with long duration data. Structural acoustics problems often have just the opposite characteristics with irregular spacing of multicomponent sensors that are in inhomogeneous wave fields with transient data. These eliminate many array processing methods and require generalization of others. Recently, array processing methods such as matched field and matched mode, slowness-time spectra, and singular value decomposition have been advanced and used for structural acoustics. This presentation will give an overview of advanced array processing methods with an emphasis on those applicable to problems in structural acoustics. [Work supported by ONR.]

10:15

6SA5. Experimental measurements of wave propagation on submerged shells. Earl G. Williams (Naval Res. Lab., Code 5137, Washington, DC 20375-5000)

Various signal processing techniques, combined with cylindrical near-field acoustical holography, are used to study wave propagation on thin, endcapped cylindrical shells excited with a point driver, and submerged in an external fluid. The upper frequency limit was doubled compared to previous experiments reported in previous meetings of the ASA. In this new frequency regime ($2 < ka < 4$) k - ω processing of the data reveal that an entirely new spectrum of supersonic helical waves is excited. Comparisons with infinite shell theory reveal that these waves are the well-known shear and extensional modes of propagation. These dominate the radiation to the far field whereas the subsonic helical waves due to bending are no longer important. The acoustic pressure in the extreme near field is used along with an assortment of k -space signal processing techniques to study the propagation on and radiation from these waves on the shell. It is shown how direction filtering in k space of helical wave components can be used to reconstruct the z - t domain representations of these filtered components, and the resulting wave propagation is recorded in a videotape display. With this display the group velocity and other physical parameters can be determined experimentally.

Contributed Papers

10:40

6SA6. On wave-vector filter analysis of turbulent flow. R. H. Mellen (Kildare Corp., 95 Trumbull St., New London, CT 06320)

Convective turbulence models are based mainly on measurements of space/frequency correlation along the longitudinal and transverse axes and the wave-number/frequency spectrum is derived by Fourier transformation of an analytic model. It has been found that the low wave-number "convective ridge" resulting from the Corcos model is apparently an artifact of the rhombic Fourier window, since it does not occur with the smoother elliptic window [R. H. Mellen, J. Acoust. Soc. Am. **88**, 2891-2893 (1990)]. Wave-vector filter arrays are also used for determining wave-number/frequency statistics in this region. Relative effects of the two window models on the analysis of experimental results are examined.

10:55

6SA7. Calculation of shock spectrum from a discretely sampled velocity time history. Peter K. Kasper, Benjamin A. Slupek, and Rosalyn F. Sawczyn (CASDE Corp., 2800 Shirlington Rd., Ste. 600, Arlington, VA 22206)

A shock spectrum corresponds to the maximum response of a mechanical resonator for a given time history excitation input. With modern computer-based instrumentation, the desired time history input is often provided as discretely sampled acceleration or velocity values. Determination of a shock spectrum requires a solution method for calculating resonator response to the transient input. Beck and Dowling [J. L. Beck and M. J. Dowling, Earthquake Eng. Struct. Dyn. **16**, 245-253 (1988)] present an algorithm for computing resonator response given a

discretely sampled input acceleration time history. The current paper extends the application to account for input excitation in terms of a transient base motion velocity. The algorithm is derived from a closed-form integration of the system response with linear extrapolation between discrete input values. Verification of the discrete velocity input algorithm is demonstrated by comparison with corresponding results using a continuous time history input excitation.

11:10

6SA8. Theory and experiment of two concentric submerged cylindrical shells coupled by the entrained fluid. Shigeru Yoshikawa (5th Res. Ctr., Technol. Res. and Dev. Inst., Japan Defense Agency, Yokosuka 239, Japan), Earl G. Williams, and Karl B. Washburn (Naval Res. Lab., Washington, DC 20375-5000)

In this paper theory and experiment for the coupled vibration of two concentric cylindrical shells (double shell) are described; the inner shell contains air, the outer shell is surrounded by water, and water exists in the annular space between them. A point force is applied to the inner shell, whose displacement produces acoustic pressure in the annular space and in turn this pressure wave drives the outer shell. The nature of acoustic field in the annular fluid and its coupling effect on the shells is investigated. Using Flügge's shell equations and the Helmholtz equation, the normal modes of an infinite double shell are calculated. The theory includes the effect of initial prestress (uniform axial compression). The results of these numerical calculations in wave-number space will be compared with the results from generalized near-field acoustical holography (GENAH) experiment on a simply supported finite double shell. The wave-number/frequency representation of this double shell will be also compared with that of a single submerged shell.

11:25

6SA9. Time-dependent wave-vector analysis of the near-field pressure of a fluid-loaded cylinder. Gerard P. Carroll (David Taylor Res. Ctr., Bethesda, MD 20084), Arthur B. Baggeroer (MIT, Cambridge, MA 02139), and Edward K. Scheer (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

The response of a ribbed cylinder to impulsive excitation was measured on a cylindrical scan surface in the evanescent near field of a cylinder and previously reported [J. A. Clark, and D. Feit, J. Acoust.

Soc. Am. Suppl. 1 84, S87 (1986)]. The present paper describes the application of a modified maximum-likelihood array processing algorithm to the spatial array data in conjunction with short time FFT processing of selected temporal portions of the transient response, to provide high resolution wave-number-frequency-time information about the field. Slowness (reciprocal phase velocity) versus time plots of the response for each frequency band reveal the time of arrival and the phase velocity of the various waves traveling past the array, which are compared to theoretically predicted dispersion curves for a water loaded infinite cylindrical shell. [Work supported by DARPA/ONR.]

11:40–11:55

Panel Discussion

THURSDAY MORNING, 2 MAY 1991

INTERNATIONAL B & C, 8:00 A.M. TO 12:00 NOON

Session 6SP

Speech Communication and Psychological and Physiological Acoustics: Speech Perception and Hearing Loss (Lecture and Poster Session)

Sigfrid D. Soli, Cochair

House Ear Institute, 2100 West Third Street, Los Angeles, California 90057

Robert C. Bilger, Cochair

Department of Speech and Hearing Sciences, University of Illinois, 901 South 6th Street, Champaign, Illinois 61820

Chair's Introduction—8:00

Invited Papers

8:05

6SP1. Factors determining speech-hearing handicap in noise. Reinier Plomp and Joost M. Festen (Dept. of Otorhinolaryngol., Free Univ. Hospital, P.O. Box 7057, 1007 MB Amsterdam, The Netherlands)

Frequently hearing impairment, particularly presbycusis and noise-induced losses, manifests itself primarily as a difficulty in speech understanding in the presence of other sounds. In many cases, the hearing-impaired listener needs a 5 to 10 dB larger speech-to-noise ratio (S/N) than normal-hearing listeners. Whereas the hearing aid is very effective in compensating for sound *attenuation* in the impaired ear, no easy remedy is available for improving the S/N significantly. As in critical listening situations every dB in S/N corresponds with 15%–20% difference in the intelligibility score for sentences, even a small loss in terms of S/N will represent a severe handicap in everyday listening situations. This will be illustrated with the results of experimental data collected by the authors and their co-workers in recent years. Various factors will be considered, including (1) speech-reception thresholds for different pathologies, (2) steady-state interfering noise versus competing speech, (3) the significance of binaural hearing, lipreading, redundancy of the speech, etc., and (4) the role of the environment (reverberation) and the effect of acoustical measures.

8:30

6SP2. Interference reduction for the hearing impaired. Patrick M. Zurek (MIT, Res. Lab. of Electron., Rm. 36-730, Cambridge, MA 02139)

Despite continuing technical improvements to hearing aids, user dissatisfaction with the benefit provided remains high. The most frequent source of complaint concerns the interfering effect of environmental background noise on speech reception. This paper will review the factors thought to be responsible for this increased susceptibility to interference as well as approaches that are being taken to improve noisy speech reception through hearing aids. Two of the factors that often contribute to poor speech reception are decreased audibility of speech sounds and loss of binaurality. Techniques that have shown promise for reducing interference include adapting the frequency response of single-microphone hearing aids to minimize spread of masking, and fixed and adaptive beamforming using microphone arrays. A summary will be given of the benefits provided by these techniques and their dependencies on acoustic conditions. [Work supported by NIH.]

8:55

6SP3. Auditory psychophysical performance without a cochlea. Robert V. Shannon (House Ear Inst., 2100 W. Third St., Los Angeles, CA 90057)

Auditory psychophysical performance has been measured using electrical stimulation of the remaining VIII nerve and of the cochlear nucleus in deaf patients. Psychophysical measures of temporal envelope processing show relatively unimpaired performance in these patients compared to normal hearing, and speech discrimination scores indicate that speech information relating to temporal envelopes can be effectively transmitted and received. This finding also indicates that the cochlea and VIII nerve may play relatively little role in the following tasks: detection of gaps, detection of modulation, recovery from adaptation, nonspectral pitch discrimination, and duration discrimination. Intensity perception is impaired with electrical stimulation: dynamic range of usable loudness is only 10 to 20 dB and intensity discrimination experiments indicate only 10 to 40 discriminable intensity levels. Frequency resolution is completely absent in electrical stimulation and can be only crudely reconstructed by multiple electrodes stimulating discrete neural segments. Psychophysical experiments on multiple electrodes indicate that patients can perceive and discriminate complex dynamic patterns of electrical activity changing in both stimulation frequency and electrode location. The combination of temporal envelope formation and the coarse "frequency" resolution provided by multiple electrodes is adequate to convey a surprising amount of speech information. [Work supported by NIH.]

9:20

6SP4. Strategies for representing speech with multichannel cochlear implants. Blake S. Wilson (Neuroscience Prog., Res. Triangle Inst., Research Triangle Park, NC 27709 and Div. of Otolaryngol., Duke Univ. Med. Ctr., Durham, NC 27710)

Various strategies for representing speech information with multichannel cochlear implants will be described, including *compressed analog* (CA), *interleaved pulses* (IP), and *continuous interleaved sampling* (CIS) strategies. Results obtained in within-subject comparisons of strategies will be reviewed. In general, these comparisons have demonstrated large differences among strategies. Recent studies with the CIS strategy recorded large individual improvements and established a new standard of open-set speech recognition among seven subjects chosen for high levels of performance with their CA processors. The CIS strategy presents brief pulses in immediate succession across electrode channels, with the pulse amplitudes for each channel reflecting the envelope of the energy in a corresponding frequency band. The high rate of stimulation on each channel is designed to improve the representation of temporal events in speech, while the use of nonsimultaneous pulses is designed to increase the salience of channel cues through elimination of current summation between channels. [Work supported by NIH, through the Neural Prosthesis Program.]

9:45

6SP5. Can we really understand speech through the skin? Mary Joe Osberger (Dept. of Otolaryngol., Indiana Univ. School of Medicine, Riley Hospital A56, Indianapolis, IN 46220)

The purpose of this presentation is to (1) present longitudinal data on the speech perception abilities of profoundly hearing-impaired adults and children who use wearable vibrotactile aids, and (2) raise issues relevant to developing improved tactile devices. Data collected to date reveal that most of the profoundly hearing-impaired subjects in this study perceived speech better when using a seven-channel vibrotactile aid than when they used a two-channel device. Even with the seven-channel instrument, the highest levels of performance were limited largely to discrimination and identification of speech features (segmental and suprasegmental) and enhanced speechreading. Whereas these are clinically significant findings, it is not clear if these results reflect perception of linguistically relevant units of speech or merely perception of acoustic events. Understanding words in sentences without visual clues would provide the most convincing evidence

that speech can be understood through the skin. This level of performance has not been realized with current tactile aids. The ability of Tadoma users to understand conversational speech from feeling the articulatory movements of the talker suggests that speech understanding might be possible if devices delivered a richer speech signal to the user. This and other issues related to device development will be discussed.

10:10

6SP6. Issues in evaluating wearable multichannel tactile aids. Janet M. Weisenberger (Dept. of Speech and Hearing Sci., Ohio State Univ., Columbus, OH 43210)

The viability of the tactile system to convey information about speech sounds to hearing-impaired persons has been substantiated in a number of laboratory studies. In particular, the addition of multichannel tactile devices to lipreading can provide considerable additional information in speech perception tasks, as compared to lipreading alone. Further, studies of Tadoma have demonstrated the ability of the tactile system to transmit speech information even in the absence of visual input. The recent introduction of a number of wearable multichannel tactile devices has made it possible to extend the findings from laboratory studies into everyday clinical and educational settings. A number of factors must be considered in attempting to obtain results from these wearable devices in nonlaboratory settings that will equal or even surpass findings from laboratory studies. These include the level of background noise in the environment, the number of channels and speech processing strategy of the device, the nature and consistency of the training procedure employed, and the correlations between the physical stimulus and perceptual confusions. In addition, subject factors that permit one to define what makes a successful user of a tactile aid must be delineated. Each of these considerations will be discussed in light of recent data. [Work supported by NIH.]

10:35–10:50

Break

Poster Papers

All papers will be on display and all authors will be at their posters from 10:50 a.m. to 12:00 noon.

6SP7. An analysis of errors in lipreading sentences. Marilyn E. Demorest (Dept. of Psychol., Univ. of Maryland Baltimore County, Catonsville, MD 21228-5398), Lynne E. Bernstein (Ctr. for Auditory and Speech Sci., Gallaudet Univ., Washington, DC 20002), Silvio P. Eberhardt (Jet Propulsion Lab., Pasadena, CA 91109), and Gale P. De Haven (Dept. of Psychol., Univ. of Maryland Baltimore County, Catonsville, MD 21228-5398)

The long-range goal of this research is to understand the visual phonetic and cognitive/linguistic processes underlying the lipreading of sentences. Bernstein *et al.* [J. Acoust. Soc. Am. Suppl. 1 **85**, S59 (1989)] described development of a sequence comparison system that produces a putative alignment of stimulus and response phonemes for lipread sentences. Such alignments permit sentences to be scored at the phonemic level and also permit examination of the types of errors that occur. In this study the sequence comparator was applied to a database containing responses of 139 normal-hearing subjects who viewed the 100 CID everyday sentences [Davis and Silverman, 1970], spoken by a male or a female talker. Analysis of the alignments was made possible by the development of a powerful parsing program that tabulates the frequency of user-specified stimulus or response patterns and generates confusion matrices for selected portions of these patterns. To examine the impact of sentence environment, vowel and consonant confusion matrices derived from the sentences were compared to those obtained from non-sense syllables. To probe for context effects, performance on individual sentences was examined as a function of sentence, word, and syllable characteristics. [Work supported by NIH.]

6SP8. Lipreading sentences with vibrotactile vocoders: Performance of normal-hearing and profoundly deaf subjects. Lynne E. Bernstein (Ctr. for Auditory and Speech Sci., Gallaudet Univ., Washington, DC 20002), Marilyn E. Demorest (Dept. of Psychol., Univ. of Maryland Baltimore County, Catonsville, MD 21228-5398), David C. Coulter (Coulter Associates, Vienna, VA 22180), and Michael P. O'Connell (Central Inst. for the Deaf, St. Louis, MO 63110)

Three vibrotactile vocoders were compared in a training study involving aided and unaided lipreading: (1) the Queen's University/Central Institute for the Deaf vocoder, with one-third octave filter spacing and logarithmic output compression (CIDLog) [Engelbreton and O'Connell, IEEE Trans. Biomed. Eng. **BME-33**, 712–716 (1986)]; (2) the same vocoder with linear output equalization (CIDLin); and (3) the Gallaudet University vocoder designed with greater resolution in the second formant region, relative to the CID vocoders, and linear equalization (GULin). Nine normal-hearing and four profoundly hearing-impaired adults participated in the training study. Four of the normal-hearing subjects were assigned to either of two control groups, a group that received no vocoder, and a group that received the previously studied CIDLog vocoder [Brooks and Frost, J. Acoust. Soc. Am. **74**, 34–39 (1983); Weisenberger *et al.*, J. Acoust. Soc. Am. **86**, 1764–1775 (1989)]. The remaining subjects were assigned to the linear vocoders. GULin was the only vocoder significantly effective in aiding open-set sentence identification, and benefit extended to each subject who received that vocoder. [Research supported by NIH.]

6SP9. Speech perception ability of adults with multiple-channel cochlear implants, using the spectral maxima sound processor. Colette M. McKay and Hugh J. McDermott (Dept. of Otolaryngol., Univ. of Melbourne, R.V.E.E.H., 32 Gisborne St., East Melbourne, Australia, 3002)

The spectral maxima sound processor (SMSP) was designed at the University of Melbourne for use with the 22-electrode cochlear implant manufactured by Cochlear Pty. Ltd. The processor utilizes a bank of 16 bandpass filters that are assigned to 16 electrode pairs tonotopically. In each stimulation period (typically 4 ms) six electrode pairs are stimulated. The selection of electrodes and stimulation levels are determined by the six filters having the highest amplitude outputs. Speech perception results with a number of adult implantees were obtained and the SMSP's performance compared with that of commercially available speech processors designed for this implant, including the most recent MSP (multi-spectral-peak processor). These results show that, for the subjects investigated, the SMSP improved speech perception ability, in quiet and in noise. This was true both for vowels and for consonants. Speech tests used in this study included open-set sentences, open-set monosyllabic words and closed-set vowel and consonant confusion tests. The tests were administered in the auditory-alone condition.

6SP10. Analytic study of the fundamental speech skills test. Harry Levitt, Frank Eadie, and Karen Youdelman (Ctr. for Res. in Speech and Hearing Sci., Graduate School and Univ. Ctr. of The City Univ. of New York, 33 W. 42nd St., New York, NY 10036)

The fundamental speech skills test (FSST) evaluates basic speech production skills and is designed for use with hearing-impaired students. A field study was performed in which 250 hearing-impaired students (age range 6 through 19 yr) were evaluated. A series of factor analyses was performed on the data, the results of which showed distinct patterns among basic speech skills. Relationships with hearing loss, age, and gender were examined. The primary factors were found to be vowel production, consonant articulation, breathstream management, pitch control, stress and intonation, and syllabification. Similar rates of improvement with increased residual hearing were observed for almost all factors. Age effects differed among the factors. Male-female differences were observed primarily with respect to pitch control.

6SP11. Use of spectral and temporal cues in vowel identification by normal-hearing and hearing-impaired listeners. W. Van Summers and Marjorie R. Leek (Army Audiology and Speech Ctr., Walter Reed Army Med. Ctr., Washington, DC 20307-5001)

Listeners with sensorineural hearing impairments generally show deficits in frequency resolution accompanying their decreased auditory sensitivity. Poor frequency resolution may make it difficult for these listeners to distinguish among vowels with similar formant frequencies. In English, these vowels tend to form tense/lax pairs (e.g., *i*/*ɪ*, *æ*/*ɛ*) which differ in duration. The present study examined hearing-impaired and normal-hearing listeners' use of duration and formant frequency information in the labeling of synthetic CVC stimuli forming a /bit-bit/continuum. Durational and *F*₂ frequency cues to vowel identity varied systematically across stimuli. Frequency resolution at 2000 Hz (the *F*₂ region for *i*/*ɪ*) was measured using a notched-noise masking paradigm. A temporal difference limen for a narrow-band noise centered at 2000 Hz was also measured. Subjects with normal frequency and temporal resolution tended to rely primarily on formant frequency information in the vowel labeling task. However, subjects with abnormal frequency resolution but near-normal temporal difference limens made greater use of vowel duration in vowel identification. [Work supported by NIH.]

6SP12. Longitudinal changes in speech production in children with cochlear implants. Arlene E. Carney (Boys Town Natl. Res. Hospital, Omaha, NE 68131), Jill Firszt (Carle Clinic, Urbana, IL 61801), Cynthia J. Johnson (Univ. of Illinois, Champaign, IL 61820), and Edward Carney (Boys Town Natl. Res. Hospital, Omaha, NE 68131)

The purpose of this study is to determine when and how subphonemic changes in speech production lead to changes in speech intelligibility in three prelingually deafened children with poor speech production skills, following the receipt of multichannel cochlear implants. Measurements of open- and closed-set speech intelligibility were made pre-implant, and at 3, 6, 12, and 18 months post-implant. In addition, monosyllabic word tokens were analyzed perceptually with reduced aspect feature transcription (RAFT) [A. E. Carney, J. Acoust. Soc. Am. Suppl. 1 87, S89 (1990)] to determine global articulatory features. Results indicated that relatively large changes in syllable shape, vowel space, and consonant manner preceded small changes in speech intelligibility. RAFT results were displayed in matrix format for each test period. Matrices were compared to estimate which production changes showed consistent patterns over time. Results will be discussed in terms of base-line performance and expected rate of change for prelingually deafened children who received cochlear implants.

6SP13. Accounting for normal-hearing listeners' perceptions of syllable-initial /d/'s spoken by hearing-impaired talkers. Brad Rakerd, Lisa Holden-Pitt, and Sally Revoile (Ctr. for Auditory and Speech Sci., Gallaudet Univ., Washington, DC 20002)

A substantial corpus of data on normal-hearing listeners' perceptions of stop and glide consonants spoken by sensorineural hearing-impaired talkers of widely varying speech production abilities has been collected. This report concerns the authors' first efforts to measure acoustic cues to consonants in these productions and to relate cue presence to perceptual judgments. The focus was on /d/, as produced in citation-form /dVk/ syllables (*V* = /a/, /æ/), by ten severely through profoundly hearing-impaired talkers representing the midrange of speech production abilities among our hearing-impaired adults. The /dVk/ syllables, along with other syllables contrastive for the initial consonant, were presented to normal-hearing listeners who made identification judgments for the syllable-initial consonant and, at the same time, rated their confidence in each judgment. Acoustic analysis of the /dVk/ syllables generally revealed some form of measurable release burst at onset; however, the bursts were often abnormal for /d/ in either their onset characteristics, amplitude, duration, or spectrum. The strength and quality of the talkers' vocalic cues to /d/ idiosyncratically depended upon the identity of the syllable vowel. A token-by-token analysis of the perception data showed that some constellations of /d/ cues yielded relatively high identification accuracy, others yielded high confidence, but only a small subset yielded both. [Work supported by NIH and the Gallaudet Research Institute.]

6SP14. Development of phone communication aid for hearing impaired. D. Kanevsky, P. S. Gopalakrishnan, G. Daggett, C. Danis, E. Epstein, and D. Nahamoo (Speech Recognition Group, IBM T. J. Watson Res. Ctr., P.O. Box 704, Yorktown Heights, NY 10598)

The goal of this paper is to describe work on the use of TANGORA—an Automatic Speech Recognizer (ASR) that was developed by the Speech Recognition Group at the Watson Research Center—as a communication device that would allow a hearing-impaired person to communicate with hearing individuals over the telephone. In this implementation, the speech of the hearing individual is decoded by the ASR, and the output is displayed on a computer screen for the hearing-impaired person. The general usability of this system could be limited by the degradation in TANGORA's recognition accuracy due to (1) use of public toll-quality telephone lines (instead of high bandwidth, low-noise communication lines) and (2) using it as a speaker-independent system (instead of trained to recognize each user).

For these reasons, the present study was aimed at understanding the effect of decoder accuracy and knowledge about the topic of conversation on the comprehension ability of the hearing-impaired individual. The results of some successful experiments with hearing-impaired users are described and analyzed. The results indicate that the hearing-impaired user is able to understand the speech of the hearing user even in the case of low ASR decoding accuracy (around 70%). Such an accuracy level appears to be achievable under the real conditions described above.

6SP15. Frequency-dependent gain control by the minima in the temporal sound envelope. Janette N. van Dijkhuizen, Joost M. Festen, and Reinier Plomp (Dept. of Otorhinolaryngol., Free Univ. Hospital, P.O. Box 7057, 1007 MB Amsterdam, The Netherlands)

Conventional automatic gain control acts upon the average sound level, irrespective of whether there is a speech signal or not. As a result, during periods without a speech signal, background noise is amplified to levels experienced as "noisy" by the listener. This annoyance can be reduced by using the level of the temporal-envelope minima, rather than the average sound level, to control the gain: the information-bearing fluctuations typical for speech are preserved, whereas background noise, usually showing much less-pronounced fluctuations, is presented at a nondisturbing level. This study investigates the effectiveness of a four-channel AGC system in which the frequency-dependent amplification factor is automatically controlled by the envelope minima in the respective frequency channel. The reference was a condition without gain control, but with the amplification in the different frequency bands adjusted to warrant 100% intelligibility in quiet. The effect of the gain-control system on the signal appeared to be greatest in the case of stationary sounds, and smallest when a single speaker was present. Results for 10 listeners with a sensorineural hearing impairment show that, for various sounds frequently interfering in practice, with spectra that are comparable to that of the speech, the condition with gain control does not affect the speech-reception threshold in noise, but substantially reduces the subjective impression of noisiness when no speech communication takes place.

6SP16. Effect of spectral envelope smearing on speech reception in noise and competing speech. Mariken ter Keurs, Joost M. Festen, and Reinier Plomp (Dept. of Otorhinolaryngol., Free Univ. Hospital, P.O. Box 7057, 1007 MB Amsterdam, The Netherlands)

The effect of reduced spectral contrast on the speech-reception threshold (SRT) for sentences in noise, and on phoneme identification, was investigated with 16 normal-hearing subjects. The SRT increases—to about the same extent for a male as for a female voice—as spectral energy is smeared over bandwidths exceeding the ear's critical bandwidth. Phoneme identification shows that vowels are more susceptible to this type of processing than consonants. Vowels are primarily confused with the back vowels /ɔ, u/, and consonants are confused where place of articulation is concerned. In competing speech normal-hearing subjects show a 6–8 dB lower SRT for sentences than in steady-state noise, while sensorineurally hearing-impaired subjects do not [J. M. Festen and R. Plomp, *J. Acoust. Soc. Am.* **88**, 1725–1736 (1990)]. As frequency resolution may contribute to this effect, and as fluctuating interferences of speech are very common in daily situations, the extent of the threshold difference between noise and speech masker was investigated for a number of smearing bandwidths.

6SP17. Validation of a speech intelligibility test using SRT for hearing aid research. Michael J. Nilsson, Jean Sullivan, and Sigfrid D. Soli (Hearing Aid Res., House Ear Inst., 2100 W. Third St., Los Angeles, CA 90057)

Speech reception thresholds (SRT) were measured in the presence of spectrally matched masking noise using an English language speech

comprehension test that has been developed to address some shortcomings of current speech tests as measures of hearing handicap [M. J. Nilsson *et al.*, *J. Acoust. Soc. Am. Suppl.* **188**, S175 (1990)]. Following procedures used with Dutch materials [R. Plomp and A. M. Mimpfen, *Audiology* **18**, 43–52 (1979)], L_{eq} of the sentence materials was matched and then adjusted to compensate for relative differences in difficulty between sentences. Lists of 12 sentences were created with nearly equal distribution of phonemes. SRTs were measured using normal-hearing subjects with unmodified as well as high- and low-pass filtered conditions in an effort to measure the repeatability of the lists for different signal bandwidths. Results of these tests will be presented and the use of these materials in hearing aid research will be discussed.

6SP18. Perception of /d,n,l/ by impaired- and normal-hearing listeners. Linda Kozma-Spytek, Sally G. Revoile (Ctr. for Auditory and Speech Sci., Gallaudet Univ., Washington, DC 20002), and James M. Pickett (Windy Hill Lab., Surry, ME 04684)

Eighteen moderately to profoundly hearing-impaired and 11 normal-hearing listeners were studied for their use of acoustic cues for perception of /d,n,l/ in /æCæ/ tokens extracted from spoken sentences. The cues were in segments associated with the /d,n,l/ occlusions and with the transitions of the adjacent vowels. The segments' contribution to /d,n,l/ distinctions were examined from identification tests of the /æCæ/ tokens with consonant occlusion and transition segments degraded singly or in combination. Specifically, the consonant segments were deleted or replaced by a synthetic neutral segment; the transition segments were neutralized via substitution of adjacent pitch periods from the respective vowels. The results with normal-hearing listeners showed that the presence of any one of the consonant or transition segments supported a moderate to high level of correct /d,n,l/ identification, depending on the phoneme. In contrast, the hearing-impaired listeners showed subnormal identification of intervocalic /d,n,l/ which was associated with deficiencies in the use of the consonant occlusion and, generally, with greatly reduced use of the vowel transition. [Work supported by a grant from the NIDCD (NIH) and the Gallaudet Research Institute.]

6SP19. Methods for longitudinal measures of speech nasality in cochlear implant patients. M. L. Matthies, J. S. Perkell, and M. J. Williams (Res. Lab. of Electron., Rm. 36-511, MIT, Cambridge, MA 02139)

The validity of longitudinal measures of nasality in cochlear implant patients has been examined based on acoustic spectra, sound levels, and outputs of nasal and throat accelerometers. Speech materials consist of isolated utterances and reading passages. Preliminary observations indicate the following: The ratio of rms values of nasal and throat accelerometer outputs [Y. Horii, *Cleft Palate J.* **17**, 254–261 (1980)] may be influenced by: (a) variation in the relative levels of the two signals during the recommended calibration maneuver, production of a sustained /m/, and (b) substantial changes in SPL that accompany onset of "auditory" stimulation from a cochlear prosthesis. These observations raise uncertainty about using the throat accelerometer output as a reference and about the sensitivity of this kind of measure to longitudinal changes in nasality across experimental sessions. In addition, measures of harmonic and formant amplitudes from acoustic spectra may be confounded by changes in coupling to tracheal resonances that also accompany the activation of the prosthesis. These observations and additional measures and calibration strategies will be explored further. [Work supported by NIH.]

6SP20. Magnitude estimation scaling of intelligibility by experienced and inexperienced listeners. Lee W. Ellis (Dept. of Special Education Services, The Univ. of Toledo, 2801 W. Bancroft, Toledo, OH 43615) and Donald J. Fucci (School of Hearing and Speech Sci., Lindley Hall, Ohio Univ., Athens, OH 45701)

Fifteen experienced listeners and 15 inexperienced listeners provided magnitude estimation scaling responses to indicate the intelligibility of two sets of nine audiotaped speech samples. These samples consisted of three utterances made up of 17 words and contained all of the consonant phonemes of English. These words were arranged to form a set of either meaningful or nonsense utterances. Nine separate versions of both the meaningful and nonsense utterances were created by systematically increasing the number of phonemes produced incorrectly on each of the nine recordings. Results indicated no significant difference between the magnitude estimation scaling responses of experienced and inexperienced listeners. Explanations for the results of this study and the advantages of magnitude estimation scaling as a measure of speaker intelligibility are discussed.

6SP21. A simplified representation of speech for the hearing impaired. James M. Kates (City Univ. of New York, Graduate Ctr., Rm. 901, 33 W. 42nd St., New York, NY 10036)

A new approach to improving speech intelligibility in noise is being developed for the hearing impaired. The approach is based on sinusoidal modeling of speech, in which the speech waveform is divided into overlapping segments, the FFT computed for each segment, and the N -highest peaks are identified. The speech is then resynthesized using sinusoids having the amplitude and phase of the selected peaks, and the remaining spectral information is discarded. Using a small number of sinusoids results in a simplified speech signal; the most intense speech components of any given segment are reproduced, while the less intense

speech and noise components are not. The question under consideration is how the speech intelligibility varies as the number of sinusoids is reduced. Normal- and hearing-impaired listeners were asked to judge the intelligibility of sentences in quiet and at a 5-dB speech-to-babble ratio; the results to be presented compare unprocessed speech with speech reproduced using 16, 8, and 4 sinusoids. [Work supported by NIH.]

6SP22. Intelligibility, phonetic distortion, and listener consensus in the speech of hearing-impaired talkers. Richard Goldhor (Speech Communication Group, 36-511, MIT, Cambridge, MA 02139)

Recorded speech from 29 hearing-impaired and four normal-hearing speakers was reviewed by two trained phoneticians. Selected segments of the speech were judged according to 14 phonetic attributes such as consonant manner and place, vowel fronting, and breathiness. Magner intelligibility scores for the hearing-impaired speakers were also available. A feature-based perceptual distance measure was constructed and used to specify both the degree of phonetic accuracy of the speech and the level of consensus in listener judgments. Analysis of these data reveals strong direct relationships between phonetic accuracy, Magner intelligibility scores, and listener consensus. Analysis of the consensus data according to feature type indicates that listeners may perceive manner features most reliably, with stress, place, and voicing features perceived with successively less reliability (i.e., consensus). [Work supported by NIH.]

THURSDAY MORNING, 2 MAY 1991

LIBERTY B, 8:25 A.M. TO 12:15 P.M.

Session 6UW

Underwater Acoustics: Experimental Ocean Acoustics I

Stewart A. L. Glegg, Chair

Florida Atlantic University, P.O. Box 3091, Boca Raton, Florida 33431-0891

Chair's Introduction—8:25

Invited Papers

8:30

6UW1. The role of experiments understanding shallow water coherence and predictability. Harry A. De Ferrari (Univ. of Miami, Coral Gables, FL 33124) and Daniel Wormser (Florida Atlantic Univ., Boca Raton, FL 33431)

The combined effect of volume, surface, and bottom scattering and fluctuation in the propagating medium greatly complicates the prediction of fluctuations and coherence for bottom-limited acoustic transmission paths. Of these effects, modeling the bottom is the least well understood and most difficult. Wave fronts interact with the bottom and subbottom, which often have unknown and indeed unknowable range-dependent bathymetries and geoacoustic properties. Exploratory experiments even with limited environmental data, combined with propagation models, provide insight in the sorting out of the relative importance of numerous parameters. After a few bottom bounces, and at intermediate ranges, depending on frequency and other parameters, the sound field is no longer a predictable incoherent detail. Yet average properties such as pulse envelopes after long time incoherent averaging can be predicted and even used in very precise tomographic inversions. At longer ranges, pulse envelopes become unpredictable, even with good average range-dependent sound-speed profiles. And at still longer ranges, signal coherence may be breaking down, resulting in the loss of the 20 to 30 dB of processing gain necessary to see signals above noise. A number of experimental results from the Florida Straits with ray, normal mode, and PE models are examined. Many potentially randomizing factors seem to average out or have little effect, even in the very

9:00

6UW2. Wave-front fluctuations in the 1000-km SLICE89 experiment. S. M. Flatté, T. F. Duda, J. Colosi (Inst. of Marine Sci., Univ. of California, Santa Cruz, CA 95064), J. Hildebrand, W. Hodgkiss, P. F. Worcester, B. D. Cornuelle (Scripps Inst. of Oceanography, Univ. of California, La Jolla, CA 92093), B. M. Howe, J. Mercer, and R. Spindel (Univ. of Washington, Seattle, WA 98105)

A 1000-km acoustical transmission experiment has been carried out in the North Pacific in order to quantify long-range acoustic fluctuations at periods of hours to days, and to measure mesoscale and internal-wave sound-speed structure within the source-to-receiver vertical slice. Pseudo-random-sequence pulses were transmitted by a moored broadband sound source (centered at 250 Hz) to a moored 3-km vertical array of 50 receivers. Data are available in the form of approximately 1000 transmissions at 1/h (spanning nine days), with an embedded sequence of another 1000 transmissions at 6/h (spanning 21 h). The multipath arrivals form a clear expected pattern of a single, folded wave front in the earlier part of the received pulses. Each point on the folded wave front represents a specific ray trajectory through the ocean. Several hundred points are available on each of the pulses. Removal of barotropic tides and equipment-motional fluctuations was carried out by averaging techniques, yielding residuals that are dominated by ocean processes such as internal waves. Time-lagged and vertically lagged covariances of residuals show the temporal and spatial scales of variability within the pulse, which can be related to the behavior of ocean processes. Baroclinic tides and internal waves are apparent in the record.

Contributed Papers

9:30

6UW3. Relating ocean acoustic ambient noise to the ocean surface dynamics. Robert M. Kennedy (Naval Underwater Syst. Ctr., AUTEC, West Palm Beach, FL 33402-7517)

The acoustical significance of ocean surface dynamics has been documented for years. Motivation for establishing the required causal relations between these two physical processes has existed for decades. The goal has eluded investigators because of the complexity of both the acoustic and oceanographic mechanisms involved. Significant contributions have been made by a progression of laboratory measurements of various spatial scales as recently reported [H. Medwin, *J. Acoust. Soc. Am. Suppl.* 1 88, S1 (1990); L. A. Crum *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 88, S2 (1990)]. The work discussed here continues the progression in spatial scales by utilizing a relatively deep water basin (The Tongue of the Ocean, the Bahamas) in a quasicontrolled environment which made a broadband measurement (40 to 4000 Hz) of the vertical directional spectrum of the ambient acoustic field. The 1-yr database of directional spectrum is dominated by local (fetch limited) water surface conditions over a wide range of environmental conditions. While the exploratory nature of the experiment limited the amount of supporting meteorological and oceanographic measurements made, the results give direction to what are the likely independent variables of the experiment. A discussion of the key environmental variables and the consistency of the results with existing theories is given in the context of the design of a future experiment to relate the acoustic ambient with ocean surface dynamics.

9:45

6UW4. Ocean acoustics turbulence study (OATS). Louis Goodman, Diane Szargowicz (Naval Underwater Syst. Ctr., Newport, RI 02841), Stephen Letcher, John Oeschger, and Elena Balasco (Univ. of Rhode Island, Kingston, RI 02881)

Limited *in situ* measurements from high-frequency underwater acoustic echo sounders have suggested that there may be circumstances in which acoustic scattering from ocean temperature microstructure is sufficiently intense to be observable over volume reverberation due to biologies. A new laboratory program, ocean acoustics turbulence study

(OATS), has been undertaken to quantify the nature of such scattering and to compare laboratory results with a recently developed model of microstructure scattering. A buoyant plume characterized by three different types of fluid regions as a function of height, namely laminar, unstable, and fully developed turbulent, is used as the scattering field. Experiments are to be performed in the frequency range from 100 kHz to 1 MHz for scattering angles spanning from near forward to near backscatter. Results to date will be presented. Preliminary results have indicated observable acoustic scattering in all three fluid regions at 1 MHz, with the turbulent regime yielding scattering strengths at a scattering angle of 90 deg of order -80 dB in qualitative agreement with model predictions.

10:00

6UW5. Low-frequency backscattering from a submerged bubble cloud. Ronald A. Roy, Michael Nicholas, and Lawrence A. Crum (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

The results of a recent series of experiments designed to demonstrate resonance backscattering from a bubble cloud are presented. In these experiments, an ≈ 1 -m-diam bubble cloud was generated ≈ 20 m beneath the surface of a deep-water lake and insonified using a pulsed parametric-array projector. Through measurements of frequency-dependent backscattering, target strengths were determined for frequencies ranging from 500 Hz to 4 kHz. (The individual bubble resonance frequencies ranged from ≈ 1.5 to ≈ 5 kHz.) Results indicate an increasing target strength with decreasing frequency, which suggests the presence of a resonance response in the subkilohertz range. Experimental results will be discussed in light of theories describing the collective oscillations of bubble clouds and plumes. [Work supported by ONR, ONT, and AEAS.]

10:15

6UW6. Volume scattering measurements with a 12-kHz multibeam echo sounder. C. de Moustier (Marine Physical Lab., Scripps Inst. of Oceanography, La Jolla, CA 92093-0205)

Acoustic volume reverberation measurements were made with a 12-kHz sea beam multibeam echo sounder by recording quadrature samples of the echoes received on each of the system's sixteen 2.66° beams. For initial inspection, the signal amplitudes from three channels (a center channel and outer channels on either side) were displayed as a function of depth and along-track distance in a grey level image quantized to 4 bits. Deep scattering layers identifiable in these images were analyzed by integrating the echoes received on each beam over a 50-m depth slice containing a layer or set of layers and by correcting for beam pattern and time spread effects as a function of angular direction. Re-

sults from data recorded during night time periods in the Northern Pacific show a fairly consistent volume scattering picture with variations averaging about 10 dB over 4-km segments along the ship's track, and 2 to 3 dB over a few hundred meters across track. The centroid of the returns in the 50-m window was also calculated for each beam and showed an along-track wavy pattern with an amplitude of about 10 m and a "period" of roughly 120 to 140 m. These results are discussed along with applications for 3-D mapping of volume scattering and patchiness distribution. [Research sponsored by ONR.]

10:30-10:45

Break

10:45

6UW7. Laboratory measurements of forward and backscattering from striated surfaces. H. E. F. Williams, R. Kille (Dept. of Physics, American Univ., Washington, DC 20016), and T. C. Yang (Naval Res. Lab., Washington, DC 20375)

A tank experiment was devised to measure forward and backscattering from single and multiple rods distributed on the water-air interface oriented normal to the plane of scattering. The cylindrical rods made of acrylic simulate the soft half-cylindrical protuberances used by Twersky in calculating scattering [J. Acoust. Soc. Am. 22, 539-546 (1950)]. (The Burke-Twersky model has been used frequently to model under-ice scattering in the Arctic Ocean.) To measure the scattering function (including the specular reflection and backscattering) from the cylindrical rods, a new experimental setup using an array of small diameter (0.05 in.) transducers was used to receive the scattering returns at different angles simultaneously. The source is a large diameter (3 in.) transducer operating at 100 kHz, with a narrow beam ($< 10^\circ$) to minimize "leakage" into backscattering angles. With the cylindrical rod radius of 1.6 mm, the scattering corresponds to $ka = 0.67$, which is the equivalent of low-frequency (~ 20 Hz) scattering from the under-ice ridges. Preliminary results of scattering from cylindrical rods are in reasonably good agreement with the Twersky calculations. [Work supported by ONR.]

11:00

6UW8. Seismo-acoustic effect of trapped air pockets underneath a floating ice plate. Jacques R. Chamuel (Sonoquest Advanced Ultrasonic Res., P.O. Box 153, Wellesley Hills, MA 02181-5339)

Little is known about the characteristics, distribution, and role of trapped air pockets present underneath the Arctic ice cover. Laboratory ultrasonic modeling results are presented demonstrating striking effects of small trapped air pockets underneath a floating plate on broadband pulsed flexural waves. The thickness of the trapped air pocket is small compared to a wavelength. The experimental studies were conducted on ice, Plexiglas, and glass plates floating on water. The air pockets increase the flexural wave velocity causing horizontal refraction and shadow zones in the plane of the floating plate. A variety of air pocket sizes and cluster configurations have been investigated. The presence of a shallow-water pool on the top surface of the floating plate has a negligible effect on the flexural wave compared to an air pocket of the same dimensions located underneath the plate. A number of different phenomena occur causing the amplitude of the detected flexural wave to be attenuated or simplified. The new findings indicate that trapped air

below the Arctic ice cover may play a significant role in Arctic acoustics. [Work sponsored by ONR.]

11:15

6UW9. A comparative experiment on under-ice acoustic scattering in the marginal ice zone. Michael J. Buckingham (Marine Physical Lab., A-013, Scripps Inst. of Oceanography, La Jolla, CA 92093)

For several years a series of ambient noise experiments has been conducted in the marginal ice zone (MIZ) off the east coast of Greenland from a fixed-wing airborne platform. In the spring of 1991, in addition to ambient noise measurements, a propagation experiment will also be attempted, aimed at establishing the effect of scattering due to the under surface of the ice cover. Parallel to the ice edge, two lines of omnidirectional sonobuoys will be deployed over a distance of about 100 km, one line about 1 km out in the open ocean and the other 1 km within the ice field. Explosive shots will be detonated at both ends of the dual line. Being so close, the primary difference between the propagation conditions down the two lines is in the scattering from the ice cover relative to that from the open-sea surface—bathymetry and sound-speed profiles being essentially the same. To support this experiment AXBTs will be deployed along both lines of buoys, although, in view of salinity variations in the vicinity of the ice, airborne expendable sound velocimeters would be preferable. These are planned for future Arctic experiments. An airborne technique for measuring the under-ice profile would also be highly desirable.

11:30

6UW10. Acoustic backscatter measurements of suspended sediments as part of the 1988 STRESS experiment. James F. Lynch (Woods Hole Oceanographic Inst., Woods Hole, MA 02543), Thomas F. Gross (Skidaway Inst. of Oceanography, Savannah, GA 31416), Blair Brumley (R. D. Instruments, San Diego, CA 92131), and Christopher Sherwood (Univ. of Washington, Seattle, WA 98105)

As part of the 1988 STRESS (Sediment Transport Events on Shelves and Slopes) experiment, an upward looking 1-MHz backscatter sonar was deployed in 90 m of water off the California coast for a period of 2 months. Dubbed "ABSS" (Acoustic Back Scatter System), this instrument made vertical profiles of the suspended sediment from 1.5 to 27.0 m above bottom. Temporal sampling allowed both slow (every half-hour) sampling to look at the long time evolution of the bottom boundary layer and fast (0.5 Hz over 2 min) sampling to look at individual wave-induced suspension. During the course of the development, two major transport events (storms) were seen. In this talk, the ABSS measurement results, the correlation of the suspended sediment profiles

obtained to relevant environmental variables, the correlation of the acoustic data to other measurement techniques (e.g., OBS, transmissometer), and the ramifications for coastal boundary layer monitoring are discussed. [Work supported by ONR.]

11:45

6UW11. Site selection and characterization for the ARSRP scattering experiment. Jerald W. Caruthers, Fredrick A. Bowles, and Ashok K. Kalra (Naval Oceanographic and Atmospheric Res. Lab., Stennis Space Ctr., MS 39529-5004)

A fine-scale acoustic bottom scattering experiment is planned for the summer of 1993 on the Mid-Atlantic Ridge flank north of the Kane Fracture Zone as part of the Acoustic Reverberation Special Research Program (ARSRP). The site (or sites) of the detailed experiment will be 5×5 km. To support selection of the site, a reconnaissance experiment is planned for the summer of 1991. To aid in that selection we have developed four specific sites for collation of detailed physical and geological data for use in simulations of expected scattering results in both the reconnaissance and fine-scale experiments. This preselection of sites will assist in targeting the reconnaissance and interpretation of the resulting acoustic data, although they are not expected to be the final sites selected. The sites were selected to be almost completely within existing high-resolution bathymetry swaths and covered by, or very near to, a seismic line. Acoustic travel times to the surface and return of not much less than 5 s (water depths > 3400 m) were also required. The selected sites represent the following four geomorphic regimes: (a) smooth, flat, deep sedimented valley floor; (b) high, level, rough thinly sedimented plateau; (c) long, steep slope or escarpment; and (d) small-scale fractures, faults, or fissures. The four sites are within the region

bounded on the northeast by $26^{\circ}30' \text{ N}$, $46^{\circ}10' \text{ W}$ and on the southwest by $26^{\circ}00' \text{ N}$, $47^{\circ}20' \text{ W}$. Presented here are the physical and geological descriptions of the sites. [Work is supported by ONR.]

12:00

6UW12. Focused beamforming for vertical line arrays in near-field bottom scattering experiments. Jerald W. Caruthers and Howard A. Chandler (Naval Oceanographic and Atmospheric Res. Lab., Stennis Space Ctr., MS 39529-5004)

A fine-scale acoustic bottom scattering experiment is planned for the summer of 1993 on the Mid-Atlantic Ridge flank just north of the Kane Fracture Zone as a part of the Acoustic Reverberation Special Research Program (ARSRP). The site (or sites) of the detailed experiment will be about 5×5 km. The experimental design includes the use of four vertical line arrays moored near the bottom within the site. Several factors complicate the geometry of the experiment, one of which is that the scattering patch will be in the near field of the arrays. This will require the use of focused beamforming for viewing a scattering patch within the experimental site. Added difficulties arise in the fact that, within the experimental specifications, the arrays may not be straight and vertical and the bottom may not be horizontal. An algorithm has been formulated for the phase shifts required at elements along a distorted vertical line for focusing on points on a sloped flat plane for use in processing the data to arrive at a valid value for scattering strength. Presented are calculations that show effective "beam patterns" in the experimental geometry and other geometric considerations affecting the experiment and the calculation of scattering strength. [Work supported by ONR.]

THURSDAY MORNING, 2 MAY 1991

LINCOLN, 9:30 A.M.

Meeting of Accredited Standards Committee S12 on Noise

to be held jointly with the

U.S. Technical Advisory Group for ISO/TC 43/SC1 Noise and ISO/TC 94/SC12 Hearing Protection

D. L. Johnson, Chair S12

EG & G Mason Research Institute, P.O. Box 9024, Albuquerque, New Mexico 87119-9024

H. E. von Gierke, Chair

respective U.S. Technical Advisory Groups (TAGs)

for ISO/TC 43/SC1 and ISO/TC 94/SC12

1325 Meadow Lane, Yellow Springs, Ohio 45387

Standards Committee S12 on Noise. Working Group chairs will report on their progress under the plan for the production of noise standards. The interaction with ISO/TC 43/SC1 and ISO/TC 94/SC12 activities will also be discussed, with reference to the international standards under preparation. The Chair of the respective U.S. Technical Advisory Groups (H. E. von Gierke) will report on current activities of these International Technical Subcommittees under ISO. Plans for the next meeting of ISO/TC 43/SC1, to take place in Australia, in December 1991, will be discussed.

THURSDAY AFTERNOON, 2 MAY 1991

JOSEPH MEYERHOFF SYMPHONY HALL, 2:00 TO 3:30 P.M.

Session 7AA

Architectural Acoustics: Vern O. Knudsen Lecture on Concert Hall Acoustics

David Lubman, Chair

Hughes Aircraft Company, Building 675, MS R315, P.O. Box 3312, Fullerton, California 92634

Chair's Introduction—2:00

Invited Paper

2:10

7AA1. Concert Hall Acoustics—1991. Leo L. Beranek (975 Memorial Dr., Ste. 804, Cambridge, MA 02138)

Acoustical design of concert halls began with Wallace C. Sabine who developed the reverberation equation and advised on Boston Symphony Hall (1900). Little additional information was learned in the next half-century. The use of reflecting panels in large spaces was introduced in halls with acoustically adverse shapes between 1953 and 1959. Next followed an understanding of sound absorption by audiences in different densities of seating, the initial-time-delay gap, and the value of lateral reflections. The London Royal Festival Hall and New York Philharmonic Hall confirmed the importance of those findings and revealed additional design requirements for reflective panels and sound diffusion. Extensive subjective studies, both in actual halls and in laboratories with electronically produced acoustical environments, were performed in Göttingen and Berlin (Germany), England, Denmark, Japan, and Canada. Those studies, when combined, revealed five orthogonal subjective attributes of concert hall quality: reverberance, loudness, envelopment, intimacy, and bass balance. These, in turn, are related to the objective quantities: initial and subsequent reverberation times (RT); cubic volume and audience area; lateral reflections and diffusion; initial-time-delay gap and subsequent reflections in the first 50 ms after arrival of the direct sound; and the slope of the initial RT versus frequency curve between 125 and 2000 Hz. An additional important subjective attribute is related to the physically measured ratio of the reflected energy in the sound field during the first 50 ms and that after 50 ms. Concert halls built since 1975, principally in Europe, United States, and Japan, constitute a "laboratory" for confirmation of those subjective attributes and related objective parameters.

THURSDAY AFTERNOON, 2 MAY 1991

INTERNATIONAL A, 1:00 TO 5:05 P.M.

Session 7AB

Animal Bioacoustics: Session Honoring William E. Schevill

Arthur N. Popper, Chair

Department of Zoology, University of Maryland, College Park, Maryland 20742

Chair's Introduction—1:00

Invited Papers

1:10

7AB1. The ultrasound-detecting ear of the praying mantis—Form and function. David D. Yager (Dept. of Psychol., Univ. of Maryland, College Park, MD 20742)

Praying mantises have sensitive ultrasonic hearing mediated by a single ear located in the ventral midline of the body. The ear comprises two teardrop-shaped tympana facing each other in a deep, longitudinal

groove; they are separated by less than 150 μm . Each tympanum is backed by a tightly adherent tracheal sac. Sensory transduction by 30–35 sensory neurons takes place at the extreme anterior end of each tympanum. Neurophysiological and behavioral tests confirm the prediction that the auditory system lacks directionality. The advantage of a single ear in the midline is not clear, but physiological evidence suggests that location in the deep groove increases sensitivity by 4–6 dB. The midline ear of the mantis is part of an “early warning system” affording protection from echolocating bats. Tethered flight and free flight experiments have shown that the mantis begins responding behaviorally within 50 ms with full foreleg extension and abdomen dorsiflexion; a change in flight path begins in 100–200 ms. It is hypothesized that the abdominal flexion throws the flying mantis into a stall out of which it rolls into a power dive. The maneuvers are very effective: In field trials with wild bats, normal mantises escaped capture whenever attacked, while mantises deaf to the bats’ cries were almost always caught.

1:40

7AB2. Courtship communication and hearing in an African electric fish. John D. Crawford (Parnly Hearing Inst., Loyola Univ., 6525 N. Sheridan Rd., Chicago, IL 60626)

Among sound-producing fishes, *Pollimyrus isidori* is of particular interest because of its unusually large repertoire of sounds (5), and because its ears (sacculi) are specialized for sound pressure detection. Males use temporally patterned sounds for courting females, and the male’s sonic behavior is elicited by electric signals from the female. The courtship sounds are composed of trains of clicks, and inter-click interval (ICI) distinguishes the different sounds. The processing of these temporal features is being explored in the brain (mesencephalon) with single unit electrophysiology. The *P. isidori* auditory system is most sensitive to sounds in the region where amplitude spectrum for the communication sounds peaks (235 Hz), and at a distance of a meter from a sound-producing male (i.e., about 20 body lengths) many neurons would be at least 20–30 dB above threshold. Most neurons are broadly tuned ($Q_{10\text{dB}} \leq 2$), and precisely represent temporal periodicities in their phase-locked activity. A subpopulation of neurons shows an increased probability of response when the ICI is close to that of one of the courtship sounds. These neurons probably play an important role in the brain’s analysis of temporal features of communication sounds. [Work supported by NIH NRSA DC00020-02 and CDR P50 DC00293-06.]

2:10

7AB3. Structure–function relationships in the auditory system of the mustache bat. Thomas J. Park (Dept. of Zoology, Univ. of Texas, Austin, TX 78712)

The mustache bat depends on interaural intensity differences (IIDs) for localizing returning sonar signals. The inferior colliculus (IC) contains many neurons that are excited by sound at one ear and inhibited by sound at the other and are thus sensitive to IIDs. These neurons were studied to determine how their response properties and spatial receptive fields are shaped by the convergence of inhibitory inputs from lower centers. Inhibitory pathways (GABA-ergic and glycinergic) to the IC were identified by colocalizing a retrograde tracer injected in the IC and antibodies against putative transmitters. Glycinergic projections to the IC arise from the superior olive and GABA-ergic projections from the lateral lemniscus and IC interneurons. The effects of GABA and glycine were then assessed by iontophoresing agonists and antagonists of the transmitters onto individual IC neurons while recording acoustically evoked responses. Application of GABA and glycine inhibited responding, while application of GABA and glycine receptor blockers increased responding. Furthermore, both blockers were capable of expanding the neuron’s receptive field. This result suggests that inhibition in the IC functions to “fine tune” IID-sensitive neurons by narrowing their receptive fields. [Work supported by NIH.]

2:40

7AB4. Baleen whale vocal behavior: A synthesis of what we have and have not learned. Peggy L. Edds (Dept. of Zoology, Univ. of Maryland, College Park, MD 20742)

The earliest attempt to record the sounds of a baleen whale was by Schevill and Lawrence [Science 109, 143–144 (1949)]. Since that time, a growing body of literature has revealed that sound production by baleen whales is quite variable, including relatively narrow-band or tonal calls, and broadband, pulsed series with variable repetition rates or complex amplitude/frequency modulations, as well as flipper slaps, tail slaps, and “noisy” underwater exhalations that appear to be communicative. Although interest in the songs of humpback whales once dominated the field, numerous studies have documented the behavior and vocalizations of other baleen whale species. A brief review and synthesis of those studies will be provided, and the concepts of “contact” calls, “song,” “social sounds,” and individual variation will be considered. In addition, the limitations of acoustical censusing will be discussed. [Original research by the author was supported by ONR, AAUW, and the Cetacean Society.]

Contributed Papers

3:20

7AB5. Physical constraints of shallow water on acoustic communication by aquatic insects. T. G. Forrest, G. L. Miller, J. R. Zagar, and K. E. Gilbert (Natl. Ctr. for Physical Acoust. and Dept. of Biol., Univ. of Mississippi, University, MS 38677)

Frequency responses of shallow, freshwater ponds in northern Mississippi were measured. The response has a highpass characteristic with a sharp cutoff frequency due to the modal properties of the system. The cutoff frequency of the system is inversely related to the depth of water at the shallower transducer (projector or receiver). Frequencies below the first mode are nonpropagating, and the overall effect of this environment on propagation is about 50 dB. Several species of aquatic insects communicate in these shallow-water ponds using acoustic signals, and they must contend with the physical constraints imposed by the system. Data on the calling song (long range) of a common aquatic beetle (*Tropisternus collaris*, Hydrophilidae) are presented and discussed in relation to the propagation characteristics of their shallow pond habitats. [Work supported by USDA.]

3:35

7AB6. Frequency response of the swimbladders of fish. Thomas N. Lewis, Peter H. Rogers, David B. Rogers (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332), and Steven N. Flanagan (Georgia Inst. of Technol., Atlanta, GA 30332)

The gas-filled swimbladder of a fish resonates in the ambient noise field, scattering significant amounts of acoustic energy. This characteristic scattered field is thought to assist a fish's own hearing and also may allow for the detection and identification of other scatterers by the receiver [P. H. Rogers *et al.*, J. Acoust. Soc. Am. Suppl. 1 **85**, S35 (1989)]. The frequency response of the swimbladder can be measured *in vivo* by a noninvasive vibration amplitude measurement system [M. Cox and P. H. Rogers, J. Vib. Acoust. Str. Rel. Des. **109**, 55-59 (1987)]. The response for a variety of fish of the species *Carassius auratus* (common goldfish) and *Astronotus ocellatus* (oscar) was measured and correlated with respect to fish size. One aspect of the results further examined was the appearance of twin peaks in the response of some goldfish. X rays of the subjects indicated that differences in size of the anterior and posterior chambers of the swimbladder may be responsible. [Work supported by ONR.]

3:50

7AB7. Neural network simulation for ranging mechanism of FM-FM cortical neurons in FM bats. Zhen-Biao Lin, S. K. Chittajallu, S. Kayalar, D. Wong, and H. O. Yurtseven (Indiana University-Purdue University at Indianapolis, 1201 E. 38th St., Indianapolis, IN 46205)

A multilayer neural network is designed for modeling FM-FM neurons in the auditory cortex of FM bats. These neurons play a critical role as a "delay-dependent multiplier" to perform cross correlation. Data from FM-FM neurons in *Myotis lucifugus* were used to train different artificial neural networks. Results show that one hidden layer is adequate for modeling biological FM-FM neurons. More hidden layers give little improvement in precision, but the cost of computation

grows linearly with the number of synaptic connections. Very large networks and very high-training cycles lead to the memorization of the training patterns but result in a poor prediction for testing patterns. A neural network model with one hidden layer (15 nodes) is proposed. The results from three trained models showed that for testing patterns not included in the training set, the prediction error was small (MSE = 0.007). The dynamic behavior of the delay-dependent multiplier (described by the contour of 70% of the maximum value) is reasonable from a hunting behavior standpoint. In the search phase (low-repetition rates), a high-output sensitivity and a very wide delay window is necessary for detecting a target at an unknown distance. After detection, the repetition rate increases and the delay width drops dramatically corresponding to focusing to a specific target distance.

4:05

7AB8. Echolocation signals of the smallest odontocetes. Whitlow W. L. Au (Naval Ocean Systems Ctr., P.O. Box 997, Kailua, HI 96734)

Echolocation signals of the smallest odontocetes, *Cephalorhynchus commersonii*, *Cephalorhynchus hectori* (genus *Cephalorhynchus*), *Phocoenoides dalli*, *Phocoena phocoena*, and *Neophocoena phocoena* (family phocoenidae) are compared with those of larger dolphins. Available data seem to indicate that two distinct classes of signals exist. Echolocation signals of *Tursiops truncatus*, *Pseudorca crassidens*, and *Delphinapterus leucas* studied at the Naval Ocean Systems Center have peak frequencies between 100-120 kHz, with high amplitudes (210-225 dB re:1 μ Pa), short durations (50-70 μ s), and wide bandwidths (30-40 kHz). The smallest odontocetes tend to emit signals having peak frequencies between 120 and 140 kHz, with low amplitudes (< 170 dB re:1 μ Pa), long durations (170-430 μ s) and narrow bandwidths (7-11 kHz). Double pulses are also emitted regularly by the smaller odontocetes and very infrequently by the larger dolphins. The signals used by the smaller animals may reflect an adaptation resulting from constraints associated with their small size and may also reflect differences in the generation mechanism. Some of the properties of high-frequency, narrow-bandwidth biosonar signals used by the smaller odontocetes will be discussed, along with comparative advantages and disadvantages of these signals.

4:20

7AB9. Acoustic response times (RTs) for *Tursiops truncatus*. S. H. Ridgway, D. A. Carder, P. L. Kamolnick, D. J. Skaar (Biosciences Div., Naval Ocean Systems Ctr., San Diego, CA 92152), and W. A. Root (Dept. of Math. C-012, Univ. of California at San Diego, La Jolla, CA 92093)

Seven dolphins (4 males and 3 females age 5-30+) were trained to make underwater acoustic responses (ARs = whistles or pulse trains) to tonal or click train (10-ms interclick interval) stimuli (St). After first training, St delivery and AR and RT recording was computer controlled. St duration varied from 60-450 ms. St were 120 dB (re:1 μ Pascal peak to peak at 1 m) \pm 24 dB in 6-dB steps. With the dolphin at 1-m depth and 1 m from St hydrophone, the trainer started a randomly variable 3- to 20-St block. The computer selected St from a file in random sequence and interval (1.1-2.1 s in 0.1-s steps) and offered St

via a St generator. ARs were received by another hydrophone, digitized, and stored. Each AR file with 20–200 St was edited on a CRT display of a 700-ms St window. No-Ar trials, noisy trials, and wrong ARs were identified. RT varied with individual, and with amplitude and duration of St. Median RT typically was less than mean by 1%–5%. Median simple RT (1 St, 1 AR) ranged from 145 ms to just over 300 ms. Median choice RT (2 unlike random St, 2 unlike ARs) ranged from 170–448 ms.

4:35

7AB10. Sounds recorded in the presence of sei whale, *Balaenoptera borealis*. Amy Knowlton, Christopher W. Clark, and Scott Kraus (Cornell Lab. of Ornithology, 159 Sapsucker Woods Rd., Ithaca, NY 14850)

Opportunistic recordings were made on 12 days during the months of August and September in 1986–1989 between Browns and Baccari Banks on the Nova Scotian shelf. Sei whales, *Balaenoptera borealis*, were seen in the recording area on 16 of the 32 recording sessions. Subsequent signal analysis revealed that a distinctive sound type was recorded on all of the 16 recording sessions when sei whales were seen and on six sessions when no sei whales were seen. On some recordings two to three different sources (based on sound quality and received levels) are apparent. All these sounds share the same basic characteristics. The signal consists of two phrases, each lasting about 0.5–0.8 s, with an inter-phrase interval of 0.4–1 s, and restricted to the 1.5–3.5 kHz band. Each phrase is composed of a series of 10–20 FM sweeps lasting approximately 30–40 ms/sweep. These signals are similar to the

few sei whale sounds reported by Thompson *et al.* [in *Behavior of Marine Animals*, Vol. 3, Cetaceans, edited by H. E. Winn and B. L. Olla (1979)]. These sounds are distinctly different from those for all other cetaceans known to frequent these waters.

4:50

7AB11. Chaos in a bioacoustic signal. Edward R. Buchler, Nanda K. Alapati, and Gregory A. Mack (Radix Systems Inc., 2 Taft Ct., Rockville, MD 20850)

Zebra finches (*Poephila guttata*) emit sexually dimorphic, multiharmonic distance calls. These calls exhibit deterministic chaos. This was demonstrated by first digitizing the time series of the calls of five adult birds of each sex, then embedding a stationary portion of each, selecting an appropriate time delay (τ) value, and generating three-dimensional phase plots. From these plots, strange attractors were reconstructed. The morphology of these attractors is different for each bird and highly repeatable across calls for a given individual. Correlation dimension and local intrinsic dimension values were also calculated and support the conclusion that the signals are chaotic. The avian vocal tract has been modeled as a nonlinear system [J. H. Brackenbury, *J. Theor. Biol.* **81**, 341–349 (1979); N. H. Fletcher, *J. Theor. Biol.* **135**, 455–481 (1988)]. Its nonlinear dynamics are the probable cause of the chaotic properties of the signals. It is suggested that chaos may be a common property of bioacoustic signals produced by systems which utilize fluid forces to generate sound. It has not yet been determined if the birds can distinguish between sounds based on their chaotic properties.

THURSDAY AFTERNOON, 2 MAY 1991

LIBERTY A, 1:00 TO 4:10 P.M.

Session 7EA

Engineering Acoustics: Quiet Product Design

David L. Bowen, Chair

RH Lyon Corporation, 691 Concord Avenue, Cambridge, Massachusetts 01238

Chair's Introduction—1:00

Invited Papers

1:05

7EA1. Designing quiet products—An acoustical overview. Richard H. Lyon (Dept. of Mech. Eng., MIT, Cambridge, MA 02139)

Product design is a multidisciplinary activity that combines scientific and technical understandings with product requirements and manufacturing process constraints to achieve an acceptable prototype. Within the design process there is a continuing interaction among all the requirements, such as structural integrity, lubrication, durability, appearance, etc. For acoustical goals to be achieved in this process, acoustical constraints and principles must be represented in the tradeoffs and compromises that are necessary in design. A major question is how these acoustical aspects are to come to the bargaining table. Very few design engineers understand acoustics deeply enough to know what is critical and what is not to meet product requirements, and very few noise control engineers have broad design experience. The author's own experiences in this area will form the basis for a discussion of these issues.

7EA2. Prospects for low-noise, high-performance chain saws: An achievable goal? S. E. Keith (Institute for Microstructural Sci., Natl. Res. Council, Ottawa, Ontario K1A 0R6, Canada and Dept. of Mech. and Aeronaut. Eng., Carleton Univ., Ottawa, Ontario K1S 5B6, Canada) and A. J. Brammer (Inst. for Microstructural Sci., Natl. Res. Council, Ottawa, Ontario K1A 0R6, Canada)

The noise of chain saws powered by internal-combustion engines depends on the relative strengths of four major sources; engine exhaust and intake, noise radiated from the structure associated with engine operation (so-called "mechanical" noise), and noise associated with the cutting process. The prospects for developing low-noise, high-performance saws have been explored by rank ordering the source strengths and examining the potential for reducing the sound pressure produced by the most significant source, or sources, at each frequency. At the present state of development, a further reduction in the noise at the operator's ear of approximately 5 dB can be expected by improving the exhaust and intake mufflers of a typical 3-kW saw, but the A-weighted sound level will remain in excess of 100 dB. Control of the mechanical noise will be required to reduce the A-weighted sound level experienced by the operator to 90 dB, within the constraints of minimal increases in weight, size, and cost, and maintenance of mechanical efficiency.

7EA3. Experimental design of centrifugal fans for minimum noise generation. J. Stuart Bolton and Peter Konieczny (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907)

The level of broadband noise generated by airflow through a centrifugal fan has been found to depend on a variety of housing design parameters, e.g., scroll development angle and length, axial and radial inlet clearances, and cutoff clearance. In the absence of complete theories describing the effect of these parameters on the noise generation process, it has been necessary to identify quiet fan designs experimentally. The comparison of noise levels generated by fans having different designs is, however, complicated by the fact that design changes affect a fan's aerodynamic performance as well as its sound power level. Thus it is impossible in general to operate two different fans at the same operating point and thus to establish directly which fan design will result in the least noise generation while fulfilling a specified pumping requirement. In this paper, a procedure is described that allows the effects of aerodynamic performance differences to be scaled out of measured sound power levels, thus allowing fans having different aerodynamic performance to be compared and, in turn, making it possible to establish unequivocally which of several possible designs is the quietest. The procedure will be illustrated through the experimental identification of the optimum housing design parameters for a fan having an impeller diameter of approximately 0.1 m. In addition, it will be shown how a low-noise design established in this way may be physically scaled to meet a range of flow requirements, and how the noise generated by any fan within the design family may be predicted.

7EA4. Minimization of radiated sound power from structures using shape techniques. G. H. Koopmann and J. S. Lamancusa (Dept. of Mech. Eng., Penn State Univ., 157 Hammond Bldg., University Park, PA 16802)

A strategy for the optimal design of quiet structures will be presented and applied to some simple examples to illustrate the approach. One study involves the radiation of a baffled, rectangular, isotropic, flat plate of nonuniform thickness distribution. The plate's mode shapes and vibration response to a prescribed loading are calculated using finite element analysis. The radiated sound power is determined via a Rayleigh integral formulation. The optimal thickness distribution of the plate that produces a minimum radiated power condition is determined with shape optimization techniques. Four different optimization strategies are compared: (a) the minimization of radiated sound power at a single frequency, (b) the minimization of radiated sound power over a broad frequency band, (c) the minimization of the sum of modal radiation efficiencies, and (d) forcing the plate to vibrate as a "weak radiator."

7EA5. Mechanisms of noise generation in semihermetic, twin-screw compressors. J. D. Jones and R. W. Andrews (1077 Ray W. Herrick Labs., Purdue Univ., West Lafayette, IN 47907-1077)

The noise radiation characteristics of a 70-ton semihermetic, twin-screw compressor are examined under controlled laboratory conditions to identify the mechanisms of noise generation. Results indicate that a majority of the acoustic radiation occurs at the first six harmonics of the fundamental screw frequency of refrigerant gas compression and discharge. Compressor sensitivity studies are performed to correlate changes in the noise radiation characteristics and internal gas pulsations with variations in select operating parameters of the compressor. Sensitivity tests suggest that the primary mechanisms of noise generation are discharge gas pulsations, rotor chatter, and flow noise. Discharge gas pulsations, which are created by a

mismatch between the chamber pressure and discharge pressure, are significant at all capacities. At low capacities, the reduced refrigerant flow rates and oil circulation decreases the gas cushion between the twin rotors, consequently promoting rotor chatter. Conversely, at high capacities the increased refrigerant flow rate and oil circulation reduces the rotor chatter effect but introduces flow noise. [Work supported by United Technologies Carrier.]

3:10

7EA6. Acoustic diagnostics of a HVAC system. G. C. Lauchle, T. A. Brungart, and J. Tichy (Grad. Prog. Acoust. Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804)

The heating ventilation, and air conditioning (HVAC) system of an automobile can be a source of acoustic annoyance, particularly when it is operated under maximum airflow conditions. In order to suggest possible design changes for the purpose of noise control, a systematic characterization of the acoustic sources and mechanisms must be conducted initially. This presentation addresses such a characterization for a typical system. The approach is based on acoustic intensity measurements of the stand-alone HVAC system operating under maximum airflow conditions in a flow-through anechoic chamber. The experimental data indicate that the centrifugal blower is the dominant low-frequency source of noise, while separation zones and the flow over sharp edges within the HVAC system ducting are secondary sources of noise that become increasingly dominant as the frequency increases into the kHz range. Qualification of these identified sources of sound is aided by detailed flow visualizations of the subject system. [Work supported by Ford Motor Co.]

3:25

7EA7. Axial fan noise reduction by improved inlet conditions. K. Anthony Hoover (Cavanaugh Tocci Assoc. Inc., 327 F Boston Post Rd., Sudbury, MA 01776)

Low-frequency noise levels in occupiable space near an axial supply-air fan were significantly higher than expected, even though massive and correctly installed ductwork lagging was in place. It was discovered that opening a small access panel in the fan-intake plenum clearly reduced low-frequency noise. A baffle was constructed and adjusted within the plenum to distribute air more evenly across the fan inlet, thereby reducing noise levels by 10 dB in the 31-Hz octave band. Related experiences highlight the importance of proper fan-intake conditions to minimize fan-generated noise.

3:40

7EA8. Vibration control of a rotating arm using an ultrasonic motor. Zhen Wu, Xiaoqi Bao, Vijay K. Varadan, and Vasundara V. Varadan (Dept. of Eng. Sci. and Mech., Res. Ctr. for the Eng. of Electron. and Acoust. Mater., Penn State Univ., 149 Hammond Bldg., University Park, PA 16802)

Unlike an electric motor, the ultrasonic motor produces torque through ultrasonic vibrations. It has some unique advantages over electric motors such as light weight and small volume; no magnetic noise; and low-rotation speed without gears. Also, because it is driven by the friction force between the stator and the rotor, it has a very quick response time which is a highly desired feature in a control system. The motor control characteristics including the response time for speed and position control are presented. Experiments have been performed on a simple robotic system with a rotating arm. The motor can be used to control the arm location as well as actively control the residual vibration of the arm tip.

3:55

7EA9. Optimum structural configuration of a baffled beam for a minimum radiation condition. Koorosh Naghshineh and Gary H. Koopmann (Ctr. for Acoust. and Vibration, Penn State Univ., 157 Hammond Bldg., University Park, PA 16802)

A method has been devised for designing baffled beams that radiate sound inefficiently. The problem has been broken into two steps. First, given the overall length of the beam, a velocity distribution is found that would result in a beam that radiates sound least efficiently. This particular velocity distribution is referred to as the "weak radiator" mode shape. Second, given a choice of material properties, the structural configuration of a beam is found that exhibits the weak radiator mode shape. In the first step, a finite element adaptation of the Rayleigh integral has been employed in conjunction with the Lagrange multiplier theorem to obtain a beam velocity distribution that minimizes the radiated sound power. In the second step, extensive use of finite element modeling, as well as linear programming techniques, has been made. The result is the optimum structural configuration of a beam that exhibits a weak radiator mode shape as a natural mode shape.

Session 7PA

Physical Acoustics: Propagation and Radiation

Victor Sparrow, Chair

Graduate Program in Acoustics, Pennsylvania State University, 157 Hammond Building, University Park, Pennsylvania 16802

Contributed Papers

1:00

7PA1. Reflection and transmission of spherical waves incident on a concentric spherical interface. David T. Blackstock (Appl. Res. Lab. and Mech. Eng. Dept., The Univ. of Texas at Austin, Austin, TX 78713-8029) and Christopher L. Morfey (The Univ. of Texas at Austin, Austin, TX 78713-8029 and Univ. of Southampton, Southampton, Hampshire SO9 5NH, England)

The traditional analysis of sound reflection and transmission at an interface between two lossless fluids is for plane waves and a plane interface. Considered here is reflection and transmission of a spherical wave at a concentric spherical interface, either concave or convex. The pressure reflection (R) and transmission (T) coefficients are found to be complex; the coefficients for a convex interface are complex conjugates of those for a concave interface. Among the (somewhat surprising) results are these. First, although at high frequency the expressions for R and T are the same as for plane waves, at low frequency $R \rightarrow -1$ regardless of the value of the ratio $\rho_2 c_2 / \rho_1 c_1$ (provided $\rho_2 \neq \rho_1$). Second, perfect transmission requires both $\rho_2 = \rho_1$ and $c_2 = c_1$, not just $\rho_2 c_2 = \rho_1 c_1$. Third, if the source is a monopole and the interface radius $a \rightarrow 0$, the sound power transmitted into the second medium is that expected for a single fluid; the same is not true, however, if the source is a dipole. [Work supported by ONR, NASA, and Southampton University.]

1:15

7PA2. Effect of dispersion on a plane ultrasonic pulse. Christopher L. Morfey¹ (Inst. of Sound and Vibration Res., Univ. of Southampton, Southampton SO9 5NH, U.K.)

Propagation through typical biological media involves dispersion as well as attenuation, since the attenuation coefficient varies with frequency more slowly than f^2 in the ultrasonic range. Dispersion effects are explored for plane pulses in a homogeneous medium. The smoothing of an initially steplike pulse as it propagates through different attenuating media is illustrated numerically. Pulse waveforms are compared with and without allowance for dispersion, using a family of model $\alpha(f)$ curves. The associated dispersion in each case is obtained from the Kramers-Kronig relations, which ensure that the propagation transfer function is causal. ¹On leave 1990-91 at Pennsylvania State Univ., Ctr. for Acoust. and Vib. and Univ. of Texas at Austin, Appl. Res. Lab., Austin, TX 78713-8029.

1:30

7PA3. Measuring the frequency-dependent ultrasonic attenuation of liquids with a pulse transmission method. Peng Jiang and Robert E. Apfel (Dept. of Mech. Eng., Yale Univ., P.O. Box 2159, New Haven, CT 06520)

To measure the ultrasonic attenuation of a lossy liquid or liquid mixture, an experiment setup using a pulse transmission method has been developed. The setup consists of a focused transmitting transducer, a measurement cell, which is used to hold the liquid, a PVDF receiver, a preamplifier, and a digital scope. It has several advantages over other methods usually employed in attenuation measurement. Compared with $\alpha = 0.109 f^2$ measured by ultrasonic interferometer [G. S. Verma, J. Chem. Phys. **18**, 1352-1354 (1950)], the attenuation of olive oil is found to be fitted by either $\alpha = 0.103 f^{2.013}$ or $\alpha = 0.111 f^{1.953}$, where α is the attenuation in dB/cm and f the frequency in MHz. The error sources in the method are analyzed both theoretically and experimentally. The attenuation of some liquids and liquid mixtures is given. The potential use of the setup in industry and medicine is discussed. [Work supported by the National Institutes of Health through Grant No. 5R01CA39374.]

1:45

7PA4. Acoustic Bloch wave energy transport and group velocity. Charles E. Bradley and David T. Blackstock (Appl. Res. Lab., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

The relationship between the steady-state rate of energy transport and the group velocity is investigated for acoustic Bloch waves in a periodic waveguide. A time-average energy flux relation is derived and used to find the energy transport velocity for an arbitrary periodic waveguide. An apparent disparity between the energy transport velocity and the power delivery is discussed. The group velocity is derived using a Bloch wave generalization of the usual Fourier transform method and is shown to be equal to the rate of energy transport. The integral transform method works well for the boundary value problem as the associated Bloch wave transform is relatively straightforward. The initial value problem, however, involves the *inverse* Bloch wave transform, the problems associated with which are discussed. [Work supported by the Office of Naval Research.]

2:00

7PA5. Surface roughness induced attenuation of ultrasonic waves transmitted through a liquid-solid interface at oblique incidence. Peter B. Nagy (The Ohio State Univ., 190 W. 19th Ave., Columbus, OH 43210)

Scattering of ultrasonic waves at a randomly rough liquid-solid interface has been studied for a long time and continues to be of considerable interest. Most previous efforts were directed at studying the diffuse reflection from rough surfaces while much less is known about the

transmitted component that plays a very important role in ultrasonic nondestructive evaluation of material properties. A comprehensive experimental study has been conducted to measure the surface roughness induced attenuation of ultrasonic waves transmitted through liquid-solid interfaces at oblique incidence. Sand and shot-blasted aluminum samples with 5- to 50- μm rms roughness were used to determine the frequency spectrum of the transmitted beam between 2 and 20 MHz. A simple first-order phase perturbation technique [Nagy and Adler, J. Acoust. Soc. Am. 82, 193 (1987)] was extended for oblique incidence to derive approximate analytical formulas for the frequency and angular dependence of the surface roughness induced attenuation of the coherently transmitted field. Good agreement was found between these analytical predictions and the measured experimental data. Further improvements are expected from comparing the data to more sophisticated theoretical techniques adapted to the case under consideration. [Work supported by NSF under contract ECD-9008272.]

2:15

7PA6. On the liquid-wedge technique to generate Scholte-Stoneley waves. F. Luppé (LAUE, U.R.A. CNRS 1373, Université du Havre, Place R. Schuman, 76610 Le Havre Cedex, France), J. Doucet, and M. de Billy (GPS, Université Paris 7, France)

The liquid-wedge technique is described in the case of the generation of a Scholte-Stoneley wave on a water-aluminum interface with the use of an alcohol wedge. The exact conditions under which the desired Scholte-Stoneley wave is actually generated are reviewed. In particular, special attention has to be given to the selection of the incidence angle given by the Snell-Descartes laws, the relative position of the transducer with respect to the wedge, and the hydrostatic condition relating the alcohol height to the one of water, which ensures a good coupling. Measurements are presented, and the damping of a 2-MHz Scholte-Stoneley wave propagating over a 400-mm distance on a water-aluminum interface is investigated and compared to theoretical calculations. Results of a theoretical study, done by L. Sebbag, of the reflection coefficient of an either homogeneous or evanescent plane wave on an alcohol-aluminum-water interface are also presented. They confirm the actual generation of the Scholte-Stoneley wave at the water-aluminum interface. [This work has been done at Paris 7 University and supported by French D.R.E.T.]

3:00-3:15

Break

3:15

7PA9. Statistical description of ray diffusion in random inhomogeneous media. Alexander I. Saichev and Yevgeny G. Shchemelev (Dept. of Radiophysics, Nizhny Novgorod State Univ., Gagarin 22, Nizhny Novgorod 60332, USSR)

The ray approach is often used when coordinates and incoming angles for waves, propagating through smooth inhomogeneities, are analyzed. The probability density for coordinates and angles of rays with

2:30

7PA7. Acoustic scattering from porous aggregates. Kavitha Chandra and Charles Thompson (Dept. of Elec. Eng., Lab. for Advanced Computation, Univ. of Lowell, Lowell, MA 01854)

The interaction of acoustic waves in a two-phase biological medium characterized by randomly oriented porous aggregates is examined by direct solution of the integral equation representing the scattered and propagating field. The Born series is shown to converge in the limit as the compressibility variation is of order zero. However, pathological conditions exhibit a high degree of porosity requiring solutions to be carried out in the high compressibility limit. This results in a divergence of the perturbation series. By considering the Born series in an asymptotic sense, we obtain solutions by applying the optimal truncation rule and in extreme cases by an analytic continuation beyond the region of convergence by the method of Padé approximants. The present problem is of interest, in that angular measurements of the scattered field around the sample provides estimates of the characteristic pore sizes of the biological sample examined.

2:45

7PA8. Determination of transient acoustic wave fields in continuously layered media using the modified Cagniard method. Martin D. Verweij and Adrianus T. de Hoop (Lab. of Electromag. Res., Dept. of Elec. Eng., Delft Univ. of Technol., P.O. Box 5031, 2600 GA Delft, The Netherlands)

A combination of the Neumann series solution and the modified Cagniard method is used to derive a theoretically exact space-time domain solution for the 3-D acoustic wave propagation problem in continuously layered media. In view of the application of a time Laplace transformation with a real positive transform parameter, the convergence of the iterative Neumann scheme for an arbitrary continuously layered configuration is guaranteed, both in the transform domain and in the space-time domain. The iterates, which represent successively reflected waves, are transformed back to the space-time domain using the modified Cagniard method. Using this method, continuously layered configurations with relatively large horizontal source/receiver distances lead to an unconventional type of integration contour. In contrast to the standard angular wave number/frequency domain analysis, difficulties due to "turning rays" can be avoided. The inverse transformation procedure is demonstrated for the direct wave and the wave due to single partial reflections. Numerical results for these wave constituents are presented.

fixed ray lengths is determined by a diffusion equation. But if the solution of the diffusion equation is known it is impossible to give an answer about statistical properties of the incoming angle fluctuations at the point of reception. The cause of this is that the length of rays received at the fixed point is not known. This problem is especially important when the small angle approach is violated. The probability density determination for ray length and incoming angles in the fixed point is needed here. The derivation of this probability density is suggested for determination of the statistical properties of the rays in the receiving plane.

7PA10. Directions of degenerate plane-wave propagation in orthorhombic media. Michael Schoenberg (Schlumberger Cambridge Res., Madingley Rd., Cambridge CB3 0EL, England) and Klaus Helbig (Rijksuniversiteit Utrecht, Vening Meinesz Laboratorium, POB 80.021, 3508 TA Utrecht, The Netherlands)

At each direction of propagation in an anisotropic elastic medium, there are three possible plane waves, each with its characteristic phase slowness. Points at which one or more of the sheets of the slowness surface are in contact are "degenerate" and specify directions for which the characteristic equation has repeated roots. For an orthorhombic medium, one with three mutually orthogonal mirror symmetry planes (and no further symmetry, for the purposes of this discussion), the degenerate directions are isolated. For degenerate directions that lie in any of the symmetry planes, the characteristic cubic equation in three variables, i.e., the squares of the three components of the slowness vector, is factorable, and the problem reduces to solving for the intersection of a straight line and a conic. The problem is much more complicated out of the symmetry planes as the characteristic equation is not factorable. However, from the fact that in a degenerate direction, the Christoffel equations' 3×3 matrix of coefficients, whose eigenvalues and eigenvectors give the slownesses and polarizations of the three waves, is rank one or less, one finds three linear equations on the squares of the three components of the repeated slowness vector. If the squares of the components are all positive, there is a single degenerate direction, not lying in any symmetry plane, in each symmetric octant, and explicit expressions for all relevant quantities are readily derived; if not, there is no degenerate direction that does not lie in a symmetry plane.

7PA11. Acoustic wave propagation in a temporal and spatially varying tube. Charles Thompson and Tam Le (Dept. of Elec. Eng., Lab. for Adv. Comput., Univ. of Lowell, Lowell, MA 01854)

Acoustic wave propagation in a temporal and spatially varying tube is investigated. The method of multiple scales is used to determine the dispersion of wave packets. This primary objective is to ascertain the effect of variations in tube shape on the state transition probabilities used in hidden Markov models of speech.

7PA12. An approximate numerical solution for the general radiation problem by combining the method of wave superposition and the singular value decomposition. John B. Fahnlne and Gary H. Koopmann (Ctr. for Acoustics and Vibration, Penn State Univ., 157 Hammond Bldg., University Park, PA 16802)

In the method of wave superposition, the field due to an arbitrarily shaped radiator is written in terms of the sum of the fields due to a finite number of simple sources enclosed within the radiator. The strengths of the sources are determined by requiring that the sum of their individual fields reproduce the radiator's normal surface velocity at a finite number of locations either exactly or in the least square sense. Through the singular value decomposition (SVD), the dipole matrix, which relates the source strengths and normal surface velocities, can be written as a product of two unitary matrices and a real, diagonal matrix. Each of the unitary matrices represents a set of mutually orthogonal mode vectors, while the diagonal matrix represents the singular values associated with these modes. This decomposition is shown to contain the first N terms of the exact multipole expansion for the solution of the associated boundary value problem.

7PA13. Acoustic radiation from a chaotic viscous layer. Charles Thompson and Arun Mulpur (Dept. of Elec. Eng., Lab. for Adv. Comput., Univ. of Lowell, Lowell, MA 01854)

The Stokes boundary layer has been shown to exhibit linear instability with increasing amplitude of acoustic excitation. Special consideration is given to the amplitude range above which these disturbances bifurcate from linear stability. It is found that oscillatory modulation present in the basic-state results in successive period-doubling bifurcations for three-dimensional vortical disturbances. The acoustic radiation from these unstable vortical disturbances will be addressed.

7PA14. Application of direct integral-equation method in acoustic-wave diffraction. L. C. Huang (Code 1945, David Taylor Res. Ctr., Bethesda, MD 20084-5000), C. J. Huang, and A. T. Chwang (Dept. of Mech. Eng., The Univ. of Iowa, Iowa City, IA 52242)

The direct integral-equation method (DIEM) is applied to study the diffracted acoustic field of a point source at an arbitrary location relative to a ring aperture in a soft baffle. The velocity potential at an arbitrary field point is expressed in terms of a surface-source distribution with complex densities. A set of eight real Fredholm integral equations of the second kind is used to determine the surface-source densities. These equations are transformed into discrete forms by applying the Gauss-Legendre quadrature formula in the radial direction and the best possible numerical integration formula in the angular direction. In comparison to the boundary-element method, the DIEM is more efficient, accurate, and flexible. The advantages of DIEM become apparent, especially when the parameters in a given problem vary within a large range. The numerical result of source strengths on the baffle surface in the far field agrees very well with the asymptotic solution. The effect of different parameters on the diffracted acoustic field is systematically studied and compared. These parameters include the location of the acoustic source, the wave number, the size of the ring aperture, as well as the thickness of the baffle. The numerical results show that the baffle thickness and the size of the ring aperture have relatively little influence on the diffracted acoustic field within the tested range. However, the wave number has a significant effect on the diffracted pressure field.

7PA15. A flow-powered very low-frequency underwater tone source. S. A. Elder (U. S. Naval Acad., Annapolis, MD 21402-5026) and S. Yoshikawa (Naval Res. Lab., Washington, DC 20375-5000)

Previous studies of underwater flow excited cavity resonance in streamlined towed models have shown that large amplitude acoustic oscillations can be achieved at frequencies under 40 Hz for towing speeds in the range of 5-15 kn. The present investigation aims to develop an understanding of the phenomenon that could permit its utilization as a nonpowered low-frequency underwater tone source. A resonant cavity was constructed in the form of a rectangular box with a vertical slot cut in the side, allowing uncomplicated prediction of wall vibration and radiation. The box was then mounted in a fiberglass fairing to produce a uniform turbulent boundary layer at the location of the slot. Before towing, identification of those vibration modes that provoke flow into and out of the cavity was performed in the laboratory, using modal analysis. Early tow tank runs are showing that tonal radiation seems to occur at expected range of speeds and frequencies, though the level of the sound is less than desired. Further "voicing" of the device is underway (sharpening the edge of the slot, raising the Q of the resonator by using more resilient mounting, etc.).

Session 7PP

Psychological and Physiological Acoustics: Hearing Aids and Audiological Measurement

Lynne Marshall, Chair

Naval Submarine Medical Research Laboratory, Box 900, Groton, Connecticut 06349

Contributed Papers

1:30

7PP1. Constraints on nonlinear amplification for recruitment of loudness compensation. Apichat Tungthangthum and Janet C. Rutledge (Elect. Eng./Comput. Sci. Dept., Northwestern Univ., Evanston, IL 60208)

A nonlinear amplification technique has been developed to compensate for recruitment of loudness in sensorineural hearing losses [J. C. Rutledge, "Time-Varying, Frequency-Dependent Compensation for Recruitment of Loudness," Ph.D. thesis, Georgia Inst. of Technol., 1989]. This technique uses a sinusoidal speech model to incorporate a model of the psychoacoustic masking of sinusoids in normal hearing and in hearing-impaired persons. It operates on both a time-varying and frequency-dependent basis. The drawback is that the model processing may distort the vowel formant structure and thus cause confusions between vowels. Therefore constraints on the relative amplitude levels of the sinusoidal components at different frequency regions should be incorporated into the model. These constraints have been studied and will be discussed.

1:45

7PP2. Loudness matching for compressed speech signals. Matthew H. Bakke, Arlene C. Neuman, and Harry Levitt (Ctr. for Res. in Speech and Hear. Sci., CUNY Graduate Ctr., 33 W. 42nd St., New York, NY 10036)

The purpose of this experiment was to determine a rule for matching the loudness of uncompressed and compressed speech samples. Eight normal hearing and 16 sensorineural hearing-impaired subjects (divided into four groups of differing severity and configuration) listened to two continuous speech samples (male and female talkers) under conditions of linear and compression amplification. Compression conditions consisted of 12 selected combinations of varying compression ratio, threshold, and release time (compression ratio = 2:1, 4:1, 8:1; knee point = 5, 10, 15, and 20 dB below the highest speech peak; release time = 20, 200 ms). Subjects matched the loudness of the compressed signals to a reference uncompressed signal using an adaptive procedure. Overall levels and cumulative distributions of the processed speech signals were obtained. Estimates of equal loudness based upon rms level and 90th percentile of the cumulative distribution were both found to closely match the gain selected by subjects. [Work supported by NIH Grant No. 2PO1 DC00178.]

2:00

7PP3. Reliability, sensitivity, and validity of magnitude estimation, paired comparisons, and category scaling. Suzanne C. Purdy (Audiol.

Sec., Dept. of Physiol., Univ. of Auckland, Private Bag, Auckland, New Zealand) and Chaslav V. Pavlovic (Univ. of Iowa, Iowa City, IA 52242)

This study investigated the reliability, sensitivity, and validity of speech intelligibility judgments for hearing aid evaluation. Subjects judged the intelligibility of sentences using either magnitude estimation, category scaling, or paired comparisons. Speech recognition scores for CID sentences and NU-6 words were also obtained. Test conditions were selected using articulation index (AI) theory. AI theory predicts that speech intelligibility is monotonically related to the articulation index which depends on the available speech dynamic range and the importance $I(f)$ of each frequency band for speech intelligibility. The speech signal was bandpass filtered so that the AI increased monotonically regardless of $I(f)$. Filter conditions were selected that produce an increase in AI for speech materials spanning the range of probable importance functions, namely nonsense syllables, easy speech, and average speech [C. V. Pavlovic, J. Acoust. Soc. Am. **82**, 413-422 (1987)]. Speech scores and intelligibility judgments were obtained for each filter condition. Test-retest reliability was poorest for paired comparisons and CID sentence scores. There were no differences in sensitivity among the scaling procedures. Intelligibility judgments and NU-6 scores were more sensitive than CID sentence scores to differences among conditions. The results indicated that intelligibility judgments are valid measures of speech recognition.

2:15

7PP4. Supra-aural and insert earphone occlusion effects. Diana C. Wright and Joseph Angelelli (Dept. of Commun. Disord., Penn State Univ., University Park, PA 16801)

Monaural and binaural occlusion effects were determined for 30 subjects using mastoid placement of a B-71 bone vibrator using supra-aural earphones (TDH-49/P/N 51) and insert earphones (Eartone-3A/Earlink-3A), which had a shallow or full insertion depth. Statistically significant differences were found for earphone type, frequency, and insert earphone insertion depth. Overall, the mean occlusion effect for all conditions decreased as frequency increased from 250-2000 Hz and was greater for the binaural than the monaural occlusion and for the supra-aural than the insert earphone occlusion. Also, the mean occlusion effect was greater for the shallow compared with the full insertion depth for the insert earphone indicating the occlusion effect increased as the volume of air between the occluder and the ear drum increased. The standard deviations were not consistently higher or lower across frequency for any one occlusion condition. Clinical implications regarding the magnitude of the occlusion effect with supra-aural and insert earphone systems will be discussed.

2:45

7PP5. Normal hearing levels for an ER-3A insert earphone. Michael J. Vavrek and Tom Frank (Dept. of Commun. Disord., 5-A Moore, Penn State Univ., University Park, PA 16802)

Equivalent threshold sound pressure levels were obtained on each ear of 48 normal hearing adult subjects from 125–8000 Hz using TDH-49P earphones, referenced to a NBS-9A coupler, and ER-3A insert earphones, referenced to a HA-2 coupler. At each frequency ($N = 9$), the thresholds were significantly higher for the TDH-49P than for the ER-3A; however, the test versus retest and right versus left ear thresholds were not significantly different. The unadjusted ER-3A thresholds were in agreement with the reference equivalent threshold sound pressure levels (RETSPLs) reported by ISO [ISO 389, DAD 3, 1990]. After correcting for the subject's deviation from normal TDH-49P hearing levels and coupler (HA-2 to HA-1), the ER-3A thresholds were also in agreement with interim RETSPLs reported by ANSI [ANSI S3.6-1989, Appendix G]. A complete data base concerning normative ER-3A thresholds will be presented and discussed in reference to the results of the present study, ISO and ANSI RETSPLs, and for future standardization.

3:00

7PP6. Low- and high-frequency threshold reliability. Laura E. Dreisbach and Tom Frank (Dept. of Commun. Disord., 5-A Moore, Penn State Univ., University Park, PA 16802)

The purpose of this study was done to determine intrasubject low- and high-frequency hearing threshold reliability. Low-frequency (1, 4, 8 kHz) and high-frequency (10, 12, 14, 16, 18 kHz) thresholds were obtained on each ear of 30 normal hearing young adult subjects over four trials separated by 1 but no more than 2 weeks using a Beltone 2000 audiometer. At each frequency, the thresholds were not significantly different for the first versus second ear tested or for the four trials. Between trial threshold differences for each possible trial minus trial threshold combination ($N = 6$) were determined for each ear of each subject at each frequency. For the possible 2880 between trial threshold comparisons (6 trial combinations \times 8 frequencies \times 30 subjects \times 2 ears), only 23 resulted in threshold differences $> \pm 11$ dB. It was concluded that repeated intrasubject high-frequency thresholds were as reliable as for the lower frequencies. Clinical implications regarding high-frequency serial monitoring of hearing thresholds will be discussed.

3:15

7PP7. MacCAD, a new Macintosh-based Hypercard program for central auditory diagnostics: Description and preliminary findings. Judith L. Lauter, Nancy Solomon, and Colette Coe (Dept. of Speech and Hear. Sci., Univ. of Arizona, Tucson, AZ 85721)

Although most components of the classical auditory system lie between the periphery and association cortex, our information about auditory disorders is limited primarily to those extremes. This ignorance is

due to the fact that until recently, appropriately sensitive methodologies, both in terms of test design as well as modes for noninvasive brain monitoring, have not been readily available in the clinic. MacCAD is an attempt to address the first of these issues; a companion paper will report on results combining MacCAD with noninvasive physiological testing [repeated evoked potentials (REPs)]. MacCAD brings features of basic-research test design into the clinic, including: ease of use by both tester and client, monaural and dichotic modes for a variety of speech and nonspeech sounds, expansion capability for additional sounds, graduated difficulty for each sound set, client control of test pacing, automatic stimulus/response recording, trial-by-trial feedback, and analysis options including trial-by-trial monitoring, confusion matrices, and percent correct for individual sounds and complete sets. Initial field testing with populations predicted to have damage between periphery and language cortex, including adults with central auditory dysfunction, multiple sclerosis, Parkinson's disease, and presbycusis, indicates that MacCAD's unique features may render it sensitive to individual characteristics which, when interpreted in the context of results on other tests such as evoked potentials, may be indicative of auditory dysfunctions which are invisible to standard audiological testing. [Work supported in part by Apple Computer, Inc., Community Affairs, with the American Speech-Language-Hearing Foundation.]

3:30

7PP8. MacCAD and REP/ABRs: A new test battery for central auditory dysfunction. Judith L. Lauter and Janiece Lord-Maes (Dept. of Speech and Hear. Sci., Univ. of Arizona, Tucson, AZ 85721)

Traditionally, audiologists have specialized in the peripheral hearing system. However, recent advances in the diagnosis and treatment of peripheral problems, together with developments in brain imaging, suggest that the future of audiology is in the brain, focusing on dysfunctions resulting from CNS pathology. Financial realities dictate that the first steps toward this new future emphasize technologies which are already available and/or reasonably priced. Two components of a test battery based on these principles have been developed in our laboratories. A companion paper describes MacCAD, a Macintosh-based program for monaural/dichotic testing of speech and nonspeech sounds, which can be run on a Macintosh SE/30 with no additional hardware. Second, a simple repeated-measures modification of evoked-potential testing [repeated evoked potentials (REPs)] applied to auditory brainstem responses (ABRs) yields a dramatic increase in sensitivity to individual characteristics, based on standard EP equipment and procedures. Pilot testing with a combination of these two tests suggests that the two-part battery is not only sensitive to individual characteristics of central auditory function, but may also reveal striking correlations between the results on each test, such that details of an individual's performance on one may predict specific details observed with the other. Thus the combination of the two tests can provide complementary behavioral/physiological documentation of underlying dysfunction which may be invisible to conventional testing. In those cases where testing is available on more expensive brain-monitoring tools such as MRI, qEEG, and PET, this battery could also serve as a basis for hypothesis formulation, thus rendering their use more cost-effective.

Session 7SA

Structural Acoustics and Vibration: Structural Intensity and Power Flow

Joseph M. Cuschieri, Chair

*Department of Ocean Engineering, Center for Acoustics and Vibration, Florida Atlantic University,
Boca Raton, Florida 33433*

Chair's Introduction—2:00

Invited Papers

2:05

7SA1. On the analysis of fluid-structure interaction from the perspective of instantaneous intensity. Anthony J. Romano, Earl G. Williams, Lawrence C. Schuette (Naval Res. Lab., Washington, DC 20375), and Kevin L. Russo (Sachs/Freeman Assoc., Inc., 1401 Mc-Cormick Dr., Landover, MD 20785)

A new formulation for the structural intensity in thin shells is presented, as well as a method for determining the tangential displacement components within the shell given a knowledge of the normal displacement and surface pressure. These techniques are then applied to real, experimental data taken on a point-driven, fluid-loaded, cylindrical shell. A video will be presented that displays, simultaneously, the instantaneous structural intensity, as well as the instantaneous intensity in the fluid due to a transient pulse delivered by the point driver and the mechanism of fluid-structure interaction in the very near field will be discussed in detail from the point of view of instantaneous energy flow.

2:35

7SA2. On measurement of structural intensity in thin-walled structures. G. Pavic (Algoviceva 17, 41000 Zagreb, Yugoslavia)

In order to measure structural intensity in a thin-walled structure (shell), one has to employ intensity expressions given in terms of physical quantities that (a) are measurable and (b) refer to the external surface of the shell. These quantities are surface strains and displacements or displacement derivatives—either velocities or accelerations. Intensity expressions, which can serve as a starting basis for measurements, have been established for flat plates, circular cylindrical shells, and spherical shells. The expressions are given in terms of neutral-surface displacement and strains, which are reference quantities for any theoretical analysis and, consequently, have to be translated to the external surface domain. Intensity in a flat plate consists of two contributions: one due to extensional, the other due to flexural effects. The shell, in addition, has the influence of curvature, which couples the in-plane and the normal components of motion. As a result, large number of strains and displacement components have to be simultaneously detected in order to produce intensity readings correctly. The number of transducers required to measure these quantities is even larger, because some of the quantities are given in a differential form, implying use of multipoint finite difference schemes in practical measurements. Furthermore, some transducers, such as accelerometers, cannot measure at the body surface but somewhat above it, which necessitates corrections in readings of in-plane motions. This and other reasons (transducer cross-sensitivity, surface loading) make conventional vibration transducers impractical for this work. Preference in practical measurements should go to non-contact methods which still have to be developed. This paper describes the nature of the intensity expressions related to thin shells. Examples are given of a circular cylindrical shell in contact with an acoustic medium. Various possibilities are described for measurement of intensity using conventional transducers. Sources of measurement errors and limitations are discussed in some detail. Finally, new measurement concepts are shown, aimed at improving accuracy and reducing difficulties encountered with conventional methods.

7SA3. Flexural intensity measurements in a beam using a three-point measurement. Richard DeJong (Dept. of Eng., Calvin College, Grand Rapids, MI 49506)

A method for obtaining the flexural intensity vector in a vibrating beam has been developed using the measurement of the transverse motion at three closely spaced points. This method is superior to the two-point method in that it accounts for both the shear and bending components of the energy flow in the beam. The intensity vector can be accurately measured near discontinuities in the beam, where near-field components cause errors in the two-point method. The new method is also superior to the four- or five-point methods that use a full finite difference approximation to the flexural wave equation in that it is more compact in space and less sensitive to instrumentation errors. The new method does use an approximation that makes it sensitive to phase errors at low frequencies. The accuracy of the method and the sensitivity to phase errors are investigated using simulations and measurements on beams with discontinuities.

7SA4. The effects of fluid loading on the radiation from plane ribbed panels. G. Maidanik and J. Dickey (David Taylor Res. Ctr., Bethesda, MD 20084-5000)

The impulse response of an infinite regularly ribbed fluid-loaded panel is used to calculate the spectral response and radiation efficiency of these structures when they are excited by mechanical (line) drives. The linear response quantities, pressure and velocity, are displayed in the wave-number-frequency plane, where the patterns show many interesting features connected with the periodicity of the ribs, the mechanical damping, and the fluid loading. The combination of the linear-response quantities to form the quadratic quantities, energy and intensity, is discussed. The near-field and far-field responses in the fluid are calculated and used to determine the intensity on the panel and in the far field. These quantities are then integrated and used to determine the radiation efficiency of the panel.

Contributed Papers

7SA5. Vibrational power flow and energy localization in coupled systems. P. Ezanno (Dept. of Mech. Eng., Virginia Polytech. Inst. and State Univ., Blacksburg, VA 24061), E. K. Dimitriadis (The Catholic Univ. of America, Washington, DC 20064), and R. Burdisso (Virginia Polytech. Inst. and State Univ., Blacksburg, VA 24061)

A system of two, nearly identical Euler-Bernoulli beams, which are coupled through a torsional spring, is here analyzed in terms of power exchange between and energy localization within this nearly symmetric system. This investigation can be viewed as extending over an ensemble of systems, one member of which consists of two identical beams (the tuned state) and the rest are similar systems except for small perturbations introduced to the tuned system to create mistuned states. Harmonic excitation of members of the ensemble make it clear that vibration localization can occur for small length perturbations. Clearly, the localization degree depends strongly on the coupling strength. Here, the ensemble is examined under wide-band stationary random excitation. Computations are performed of the mean power flow between the two beams over specific frequency bands. It is clearly seen that power flow is strongest in the tuned state, but it is also significant for specific other values of perturbation. The nature of this dependence, along with its possible implications to design methodologies, such as Statistical Energy Analysis, will be discussed.

7SA6. Eigenmode statistics in large two-dimensional systems. Richard L. Weaver (104 S. Wright St., Dept. of Theor. and Appl. Mech., Univ. of Illinois, Urbana, IL 61801)

A finite difference model is used to numerically investigate the

higher-acoustic eigenmodes of membranes with random boundaries. Natural frequencies are found to distribute themselves with level repulsion and spectral rigidity like those of the "Gaussian orthogonal ensemble" in accord with the predictions of random matrix theory. These statistics have also been observed in recent experiments determining the higher eigenfrequencies of aluminum blocks. The present work goes further in that it also considers the statistics of the eigenmode *shapes*. In particular, values for the mean fourth power of modal amplitude (related to the "participation ratio" and to variances of power transfer functions in large systems) are reported, as are results on repulsion between mode shapes. [Work supported by NSF.]

7SA7. Holographic measurement of power flow in large immersed structures. Joseph A. Clark, Paul M. Honke, and J. Michael Ellis (David Taylor Res. Ctr., Bethesda, MD 20084-5000)

Simple harmonic motions of a vibrating surface can be decomposed into two component motions: a standing wave, which corresponds to a characteristic mode shape, and a traveling wave, which is associated with the flow of power along the surface. Random motions of the vibrating surface can be decomposed into a series of simple harmonic motions, each of which can be decomposed into standing and traveling components. Holographic measurement of the pressure field around an immersed, point excited, vibrating cylinder have been processed to reveal the series of standing and traveling waves noted above. The results have been visualized in computer-generated movies and used to diagnose the structural response of the cylinder and power flow along and out of its surface. In this talk, the measurement, processing, and graphic display systems used to perform the diagnosis and features revealed by the investigation will be described.

Session 7SP

Speech Communication: Issues in Production and Perception

Maureen L. Stone, Chair

National Institutes of Health, Department of Rehabilitation, Building 10, Room 6S235, Bethesda, Maryland 20892

Contributed Papers

1:00

7SP1. Effects of typicality and interstimulus interval on the discrimination of speech stimuli. Minoru Tsuzaki (ATR Auditory and Visual Perception Res. Labs., Seika-cho, Soraku-gun, Kyoto 619-02, Japan) and Jorge A. Gurlekian (Escuela de Salud Publica, Buenos Aires, Argentina)

To investigate the effects of processing time and retention interval on the perceptual judgment for speech sounds, Japanese listeners were tested with the AX discrimination task in various interstimulus intervals. Speech stimuli were synthesized to comprise both typical and atypical stimulus sets. In the typical set, the standard stimulus had characteristics of the typical Japanese /aba/ sound. In the atypical set, the standard was not a good exemplar of the Japanese /aba/ sound although it was usually recognized as /aba/. The performance of the subjects was analyzed in terms of both the center and width of discriminational processes. At relatively short intervals, there was a slight tendency for the typical standard to have a smaller width score than the atypical standard. At long intervals, there was no effect of the standard stimulus. Although the center shifted to the direction of more plosive counterparts at longer intervals, there was no effect of the standard on the center of discriminational process. The results suggest that the typical standard is represented with less noise, but this advantage decreases with increases in the interstimulus interval.

1:15

7SP2. The effects of attention on selective adaptation. Joan E. Sussman, Sandra Lane, and Valerie Lauckner (Dept. of Commun. Disord. and Sci., State Univ. of New York at Buffalo, 122 Park Hall, Buffalo, NY 14260)

Ten young adults participated in four conditions of selective adaptation preceded by baseline labeling: (1) end point-[ba], (2) "focused" (subvocal repetition of the adaptor) end point-[ba], (3) end point-[da], and (4) "distracted" end point -[da] with adaptor and test stimuli presented to the right ear and synthetic [si] and [ji] syllables presented to the left ear at 40 dB SPL for periodic labeling during presentation of the adaptor. The purpose of the investigation was to determine: (1) if the amount of adaptation could be increased by focusing subjects' attention to the [ba] adaptor and (2) if the amount of adaptation could be decreased by focusing subjects' attention away from the [da] adaptor. Results showed that (1) no significant differences between focused attention and [ba] selective adaptation (5.4% vs 3.7% shift of [ba] labels) and (2) surprisingly, an increased amount of selective adaptation for the "distractor" condition compared to the end point-[da] condition (19.3% vs 10.3% shift of [ba] labels). Results will be related to auditory versus higher-level cognitive determinants during selective adaptation labeling tasks. [Work supported by NSF.]

1:30

7SP3. Models for voicing and aspiration in stop consonants. Kenneth N. Stevens (Res. Lab. of Electron. and Dept. of Electrical Eng. and Comput. Sci., MIT, Cambridge, MA 02139)

A theoretical analysis of sound production for voiceless and voiced stop consonants has been carried out. The analysis includes the effects of active and passive (in response to intraoral pressure changes) expansion or contraction of the pharyngeal volume, active and passive changes in glottal configuration, the generation of periodic glottal vibration and aspiration noise, and the generation of transient and friction noise sources at the consonantal release. Both monopole and dipole components of the turbulence noise sources are considered. The model is used to calculate the absolute sound-pressure levels and spectra for the various components of the radiated sound for labial, alveolar, and velar stops. The predictions of the model are in reasonable agreement (within a few dB) with data from spoken syllables. Procedures for estimating the characteristics of turbulence noise sources still need to be refined, particularly the distribution of the sources downstream from a glottal or supraglottal constriction. The model can be used to infer details of the time course of supraglottal and laryngeal constrictions based on data from fine-grained acoustic analysis of utterances. [Research supported in part by NIH Grant DC00075.]

1:45

7SP4. A two-dimensional model of laryngeal flow. Fabriborz Alipour (Voice Acoust. and Biomechanics Lab., Dept. of Speech Pathology and Audiology, Univ. of Iowa, Iowa City, IA 52242) and Virendra C. Patel (Dept. of Mech. Eng., Univ. of Iowa, Iowa City, IA 52242)

Laryngeal flow was visualized by numerical simulation. Using computational fluid dynamics, a two-dimensional model of laryngeal flow was built and aerodynamic properties were calculated for steady-state laminar regime. Three configurations of vocal folds with convergent, rectangular, and divergent glottis were used to study the effects of glottal shape on the airflow. Navier-Stokes equations were solved with numerical method of Patel *et al.* A boundary fitted coordinate was used in discretizing the flow domain into exponential grids. The governing equations and coordinates were transformed into computational domain and solved with finite analytic method. Results were reported on the velocity components, pressure distributions, and wall friction coefficient at Reynolds numbers of 100-900 for different configurations. Results were compared with one-dimensional Bernoulli solutions and experimental data. It was found that flow separation exists even at low Reynolds numbers and the separation point moves toward downstream as the Reynolds number increases. [Work supported by NINCDS Grant No. NS 16320-08.]

7SP5. Glottal waveform characteristics of deaf speakers. James J. Mahshie (Dept. of Audiology and Speech-Language Pathol., Gallaudet Univ., Washington, DC 20002)

The distinctive voice quality that characterizes many deaf individuals suggests that they use phonatory adjustments for speech production that differ from normal. Little research, however, has been reported that examines the phonatory attributes of deaf individuals. The purpose of the present work was to examine glottal volume velocity waveform characteristics of deaf speakers, and to compare these characteristics to those of normal-hearing talkers. The oral airflow signals associated with non-nasalized vowels produced by five deaf and five normal-hearing subjects were inverse filtered, and the resultant signals examined. In addition to qualitative waveform descriptions, measures were made characterizing cycle-to-cycle peak volume velocity, duty cycle characteristics, and amount of unmodulated airflow (dc offset of the volume velocity signal). Result thus far reveal systematic differences from normal for some of the deaf speakers, while others, particularly those with voice quality perceived as less aberrant, had glottal waveform patterns similar to those observed in the normal-hearing subjects. Procedural issues for obtaining and analyzing the glottal waveforms of deaf individuals, and the implications of the findings of speech intelligibility and naturalness of deaf speakers, will be discussed. [Work supported by the Whitaker Foundation.]

7SP6. Acoustic analysis of essential voice tremor. Karen Colson, Mary A. Carpenter, Laura Ewy (Hearing and Speech Dept., Univ. of Kansas Med. Ctr., Kansas City, KS 66103), William Koller, and Karen Busenbark (Neurology Dept., Univ. of Kansas Med. Ctr., KS 66103)

This study examined differences in selected phonatory characteristics of essential tremor (ET) patients and normal, matched controls. All patients exhibited limb tremor of 4–10 Hz and minimal to severe audible vocal tremor. Tape-recorded speech sample for 14 ET and normal speakers from our subject pool were analyzed acoustically to determine individual and group mean fundamental frequency (f_0) measures. Both the female and male ET groups had a significantly lower mean f_0 (20–40 Hz) than that of their matched controls during sustained vowel productions. As well, the performance patterns of the ET and control groups were markedly different across repeated trials of vowel production. In connected speech, the male ET speakers' average f_0 and phonation range were significantly reduced relative to that of their matched controls. Although there was no statistical difference, female ET speakers also tended to have a reduced mean f_0 and phonation range in connected speech compared to their matched controls. These preliminary findings suggest that an abnormally low f_0 may be a critical factor

contributing to the perception of voice tremor in ET patients. [Work supported by Center on Aging, KUMC.]

7SP7. Differentiation and variability of tongue positioning in the production of German vowels. Ocke-Schwen Bohn (English Dept., Kiel Univ., Olshausenstr. 40, D-2300 Kiel 1, Germany), James E. Flege (Dept. of Biocommun., Univ. of Alabama, Birmingham, AL), Paul A. Dagenais (Dept. of Speech Pathology and Audiology, Univ. of South Alabama, Mobile, AL), and Samuel G. Fletcher (Dept. of Biocommun., Univ. of Alabama, Birmingham, AL)

Tongue–palate distances in repeated productions of German vowels were measured using glossometry. A male native speaker from northern Germany produced 10 tokens each of the 15 stressed monophthongs of German in the carrier phrase *ob er /bVp/ habe*. Tongue configurations for pairs of vowels were considered sufficiently differentiated if the mean (unsigned) distance between tongue positions exceeded 1.0 mm. For seven pairs of German vowels, tongue positions were not clearly differentiated. American English vowels, however, were produced with nonoverlapping tongue configurations in a related study [Flege *et al.*, *Lang. Speech* 29, 361–388 (1986)]. The difference between English and German may be largely due to the different mechanism used by these languages to differentiate their large vowel inventories. The overall mean s.d., associated with 10 repetitions for the 15 target positions was 0.8 mm. This measure of variability of tongue positioning is virtually the same as for American English and Spanish vowels in previous studies [J. E. Flege, *Lang. Speech* 32, 123–147 (1989)], suggesting that neither differences in inventory size nor different mechanisms used to differentiate large vowel inventories affect the precision of tongue positioning. [Work supported by NIH Grant 20963.]

7SP8. Some effects of pharyngealization on tongue shape as seen on ultrasound. Maureen Stone (National Institutes of Health, Dept. of Rehabilitation, Bldg. 10, Rm. 6s235, Bethesda, MD 20892) and Grace H. Yeni-Komshian (Univ. of Maryland, College Park, MD 20740)

The present study examined tongue shape during oral and pharyngeal consonants. A female speaker of Lebanese Arabic produced oral and pharyngealized versions of several consonants in the carrier phrase “*Kataba CV al yom*,” which means “He wrote CV today.” Ultrasound scans of the tongue were made in two planes, midsagittal and anterior coronal, during repetitions of the utterances. The [h] had the most interesting results. Pharyngealized [h] used a higher posterior tongue than oral [h]. In addition, a midsagittal humping was seen in the pharyngeal [h]. The oral [h] had a midsagittal groove as expected. Similar effects seen in other CV combinations will be discussed.

3:15

7SP9. Three-dimensional geometry of the hard palate. Kenneth L. Watkin (Biomedical Eng., McGill Univ., Montreal, Quebec H3A 2B4, Canada)

The purpose of the present investigation was to determine the three-dimensional geometry of dental casts and *in vivo* measurements of the hard palate using a pulsed inductive magnetic coil sensing/digitization device connected to an IBM PC/AT. A specially constructed palatal tracing wand, parallel port interface, and software were developed data acquisition along with 3-D database software for real time projection. Comparisons of 3-D palatal morphology will be presented. Results will be discussed in light of current anthropomorphic information on palatal shape and speech production models.

3:30

7SP10. Three-dimensional reconstruction of the palate and tongue during the production of sustained vowels. Kenneth L. Watkin (Biomedical Eng., McGill Univ., Montreal, Quebec H3A 2B4, Canada)

The purpose of the present investigation was to reconstruct the three-dimensional (3-D) shape of the oral cavity during the production of sustained vowels. Using a specially developed palatal tracing wand and ultrasonic transducer localization device, real time convex phase array images of the tongue were digitized and stored on an IBM PC/AT along with the 3-D shape of the palate. The reconstructed 3-D shapes of the tongue surface relative to the 3-D shape of the palate will be presented. Data analysis strategies will be discussed along with implications for speech articulatory modeling.

3:45

7SP11. Articulatory compensation in four-year-olds. Melanie M. Campbell (Dept. of Speech and Hearing Sci., City Univ. of New York Graduate Ctr., 33 W. 42nd St., New York, NY 10036), Richard S. McGowan, Nancy S. McGarr (Haskins Labs., New Haven, CT), and Katherine S. Harris (City Univ. of New York Graduate Ctr., NY)

The ability of six normal four-year-old children to compensate in their speech production for reduction of two types of feedback was examined. Subjects recorded repetitions of a carrier phrase containing a target word with one of three vowels (/i/, /ɪ/, /æ/) in mixed, randomly selected sequences in four conditions: normal, masking noise, biteblock, biteblock plus masking. The ability of each child to compensate was measured by comparison of the first three vowel formants of each test condition with those of the normal condition. Data from three subjects have been acoustically analyzed using LPC techniques. Preliminary results suggest that while four-year-old children more easily overcome effects of noise, they do not fully compensate for presence of a biteblock. Biteblock effects appear to vary depending upon vowel height of the target. *F2* values are lowered in biteblock production of /i/, whereas *F1* values are lowered in /æ/. Formant values for midvowel /ɪ/ are less affected. Considerable individual differences are seen in ability to compensate for feedback reduction. [Work supported by NINCDS Grant DC00121-29.]

4:00

7SP12. Perceptual evaluation of infant CV transitions. Nancye C. Roussel and Raymond G. Daniloff (Dept. of Commun. Disord., Louisiana State Univ., 163 M and DA, Baton Rouge, LA 70803)

The transition portion of the speech signal has been identified as critical to the perception of both consonants and vowels. Furui [J. Acoust. Soc. Am. 72, 43-50 (1986)] demonstrated that, for adult speakers, a 10-ms segment of the transition centered on the area of maximum spectral movement contained the most critical information for joint consonant and vowel perception and that a period roughly 50 ms long

(including the aforementioned 10-ms critical interval) was sufficient for CV syllable perception. An investigation of the perceptual cues contained within consonant-vowel (CV) transitions of infants ages 6-15 months was begun. CV-like syllables were selected from the vocalizations of two infants, a male and a female, and presented to listeners for identification. Truncated versions in which portions of the transition were systematically deleted were also presented for identification and critical intervals for the perception of consonant, vowel, and syllable were calculated and compared with those obtained for adult speakers. Comparisons across age levels were also made.

4:15

7SP13. Developmental and linguistic effects on the coarticulation of fricative-vowel productions. Kathleen A. Siren (Dept. of Speech, Commun. Sci. and Theatre, St. John's Univ., Jamaica, NY 11432) and Kim A. Wilcox (Univ. of Kansas, KS)

Previous work (Nitttrouer, Studdert-Kennedy, and McGowan, 1989) supports the notion that children attack the acquisition of speech and language in larger-than-segment units, as evidenced by greater coarticulatory interactions in young children's speech that diminish with age. In this study, the experimental design from Nitttrouer *et al.* was expanded to include familiarity versus nonfamiliarity with a stimulus item. The notion of familiarity included two factors, meaningfulness of the stimulus item, and relative amount of motor practice. Results of acoustic measurements of children's (3-, 5-, and 7-year-olds) and adult's productions confirm that children do exhibit a greater effect of a following vowel (/i/ vs /u/) on the preceding fricative (/s/ vs /ʃ/) when compared to adults. In addition, relative meaningfulness of a stimulus item appears to decrease the degree of coarticulatory interaction between these segments, regardless of age of the individual. These results are discussed with regard to traditional models of coarticulation and traditional theories of speech and language acquisition.

4:30

7SP14. Acoustic typology of infant pre- and post-linguistic protosyllables. J. Piroli, R. Daniloff, and C. J. Miller (Commun. Disord., Louisiana State Univ., 163 M and DA Bldg., Baton Rouge, LA 70803-2606)

Three hundred quasisyllabic utterances were excerpted from the vocalizations of three infants at monthly intervals in the 4- to 14th-month epoch of development. Samples were recorded while at play with the mother in a toy-filled, sound-treated chamber. *F1*, *F2*, *f0*, and *I0* trajectories for the entire quasisyllable were extracted from digital FFT/waveform displays. Syllable, consonant, vowel and transition durations, formant and *I0* envelope velocities, and jitter and shimmer measurement were made. Natural-class transcriptions and articulatory "well-formedness" evaluations were made for each utterance. A behavioral analysis checklist was available for each utterance. Analysis revealed that even perceptually labelable seemingly unexceptional tokens displayed combinations of nonadult-like acoustic parameters. Tokens were sorted by clusters of variables into a typology of emergent syllabic forms based upon adult and nonadult-like values of the classificatory acoustic-perceptual variables. The syllable well-formedness related complexly to the infant's attributed intent to communicate. The prelinguistic utterances were considerably more "juvenile" acoustically than perception would lead one to believe.

4:45

7SP15. Aging effects on the ability to use temporal context in speech perception. Marta Tetzeli and Marleen T. Ochs (Vanderbilt Univ., Div. of Hearing and Speech Sci., 1114 19th Ave. S., Nashville, TN 37212)

Elderly and young adults listened to a continuum of stimuli varying in closure duration from "rabit" to "rapid" presented at fast and normal speaking rates, with a carrier phrase and in isolation. Rates for the

phrase and word were either consistent or conflicting, to determine the relative influence of extrinsic versus intrinsic temporal cues. In addition, the effect of intensity level was examined for all the above conditions. Hearing sensitivity was within normal limits through the low-pass cutoff frequency (3.15 kHz) for the filtered speech. In contrast to previous findings with isolated words [P. J. Price and H. J. Simon, *J. Acoust. Soc. Am.* **76**, 405–509 (1984)], no significant overall age effects were seen for either the word alone conditions at the higher intensity or for any of the phrase + word conditions. These results suggest that, when hearing sensitivity is controlled, age *per se* may not explain the increased difficulty that many elderly have in processing fast speech. Follow-up research is planned to determine why these data conflict with previous findings.

THURSDAY AFTERNOON, 2 MAY 1991

LIBERTY B, 1:10 TO 5:30 P.M.

Session 7UW

Underwater Acoustics: Experimental Ocean Acoustics II

Marshall H. Orr, Chair

Office of Naval Research, 800 North Quincy Street, Arlington, Virginia 22217-5000

Chair's Introduction—1:10

Invited Papers

1:15

7UW1. Experimental measurements of acoustic propagation in the ocean. N. R. Chapman (Defence Res. Establishment Pacific, FMO Victoria, British Columbia VOS 1B0, Canada)

The comparison of experimental measurements and model calculations is an essential step in developing an understanding of the nature of sound propagation in the sea. In order to carry out such a comparison effectively, the models must be capable of producing accurate results in a range-dependent environment, and the experiment must be designed so that the geometry and environmental conditions can be reconstructed from the measured data for modeling purposes. This paper considers the question of what measurements, in addition to the acoustic field data, should be made to provide a sufficiently well-documented data set. These measurements include nonacoustic data such as navigation and ranging, source and receiver calibration and positions, and environmental data, such as sound-speed profiles, bathymetry, subbottom geoaoustics, and seafloor roughness. For range-dependent conditions, it is essential to sample the environment at a sufficiently fine degree of spatial resolution. Case studies are presented to illustrate model-data comparisons for propagation in a shallow sound channel, propagation over a sloping bottom and propagation over a thin sediment bottom. These examples are used to suggest general guidelines for designing experiments.

1:45

7UW2. Shallow-water transmission measurements taken on the New Jersey Continental Shelf. William M. Carey, James Douth^a and Lynn Maiocco Dillman (Naval Underwater Syst. Ctr., New London, CT 06320)

Calibrated acoustic transmission measurements were made under calm sea conditions on the New Jersey shelf near Amcor 6010, a surveyed area with known geophysical properties. The experiment was conducted in 73-m water with supporting measurements of salinity, temperature, and sound speed. These measurements were obtained with a vertical array of 24 equally spaced hydrophones at 2.5 m; one of which was on the bottom. A source towed at either $\frac{1}{2}$ - or $\frac{3}{4}$ -water depth transmitted one of two sets of four tones spaced between 50 and 600 Hz for each run to ranges of 4 and 26 km. The data were processed with Hankel transform and Doppler processing techniques to yield horizontal wave-number spectrum at several depths as well as mode shapes. Results were obtained along both a constant and a gradually depth varying radial. Similar modal interference patterns were observed at lower frequencies and critical angle bottom limited propagation at higher frequencies. The constant radial results were compared to calculations using several shallow-water propagation models employing both geoaoustic profiles derived from the geophysical parameters and Yamamoto's (1990) shear wave inversion. Predicted and measured levels generally agreed; however, differences in computed and measured modal interference patterns were observed. [Work sponsored by ONR 1125 OA and NUSC IR.] ^aWoods Hole Oceanographic Institution.

2:15

7UW3. Measurements of the acoustic vector wave field in the shallow ocean made by a single ocean sub-bottom seismometer (OSS). Tokuo Yamamoto (Geo-Acoust. Lab., Univ. of Miami, RSMAS, 4600 Rickenbacker Cswy., Miami, FL 33149), Thomas Nye (Univ. of Miami, Miami, FL), and Dean Goodman (Univ. of Miami, FL and Nakajima, Japan)

In the shallow oceans, acoustic waves are strongly coupled with the seabed. If the vector wave field within the seabed is measured at a point, the direction of acoustic wave propagation can be accurately determined. To test this idea, we have measured the vector wave fields that are induced by ULF/VLF (0.01–1 Hz) ambient acoustic noise and acoustic pulses (0.05–1000 Hz) generated by an airgun using a single ocean sub-bottom seismometer in shallow water 13 m deep. The maximum entropy principle is applied to the three orthogonal components of the vector field induced by the ambient noise to calculate the directional spectra. We were capable of detecting the noise field propagating from many directions that were substantiated from other independent measurements. The measured acoustic vector wave field induced by a moving source were compared with the model predictions by a WKBJ code. In the model calculations, the geoaoustic data obtained from the bottom shear modulus profiler method (Yamamoto and Trevorrow, in this conference) were used. Excellent agreements are obtained between the measured seismograms and the synthetic seismograms indicating that moving acoustic sources can be accurately determined using only two OSSs in the shallow oceans. [Work supported by ONR.]

2:30

7UW4. Laboratory scale measurements of low-frequency underwater sound propagation over a sediment layer with a hard basement. Allen J. Hundley (Science Applications Int. Corp., Campus Point Facility, San Diego, CA) and Stewart A. L. Gegg (Florida Atlantic Univ., Boca Raton, FL 33431)

Recent measurements of low-frequency sound propagation in a region where a thin sediment layer overlays a hard rock basement [Hughes *et al.*, *J. Acoust. Soc. Am.* **88**, 283–297 (1990)] showed high transmission losses at certain frequencies, which were not well predicted using a SAFARI prediction code. It was suggested that the high transmission loss levels were caused by resonances in the sediment layer. This paper will describe laboratory measurements in a similar environment using a model whose scale was $\sim 1/1000$ th of the ocean experiment. The sediment was modeled using a layer of epoxy of uniform thickness on a concrete base, and transmission loss was measured as a function of depth range and frequency. High transmission losses were found at low frequencies which have similar characteristics to those observed by Hughes *et al.* at full scale. This paper will describe the observations of the sound field at low frequencies and compare the measurements with predictions made using SAFARI.

2:45

7UW5. An experimental investigation of the horizontal refraction of low-frequency acoustic waves in shallow water. A. Yu. Shmelerv, V. G. Petnikov, A. A. Migulin (Institute of General Physics, Moscow, USSR), and James F. Lynch (Woods Hole Oceanographic Inst., Woods Hole, MA 02543)

In shallow water, the measurement of horizontal refraction is a convenient method for the study of ocean mesoscale processes. In this paper, observations of horizontal refraction over a long, fixed acoustic track in shallow water are presented. A tonal signal at 100 Hz was transmitted from a near bottom source to a long, fixed receiving array 70-km distant. The receiving array consisted of 48 hydrophones extend-

ing over 450 m horizontally. Continuous measurements of the sound-speed profile showed fluctuations of the thermocline boundary caused by long period (3.5 h) internal waves with 10-m amplitude. Simultaneously, acoustic measurements were made of the deviations from linearity of the phase front along the array. Using spatial Fourier analysis, it was seen that the variability of the phase front showed several scales. Calculations showed that long period phase fluctuations were produced by long period interval waves and short period fluctuations were produced by the interference of direct waves and waves reflected from the shore. It was also possible to account for some of the parameters of the internal waves and the reflected waves.

3:00

7UW6. Sound propagation conditions in the equatorial South Pacific Ocean. David G. Browning (New London Lab., Naval Underwater Syst. Ctr., New London, CT 06320) and Ronald N. Denham (Defence Scientific Establishment, Auckland, New Zealand)

A major part of the South Pacific Ocean is impacted by a cold water circulation induced by the Antarctic Circumpolar Current. This results in either a double or a very broad deep sound channel axis [R. N. Denham *et al.*, *J. Acoust. Soc. Am.* **81**, 787–789 (1987)]. However, nearer the equator this impact is reduced and a series of equatorial currents and counter currents come into play. An analysis is made of existing oceanographic data to determine the resulting sound-speed profile shape and sound channel axis depth. A comparison is made to profiles from the temperate regions of the South Pacific Ocean.

3:15

7UW7. Experimental detection of a slow acoustic wave in sediment at shallow grazing angles. Frank A. Boyle and Nicholas P. Chotiros (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX 78713-8029)

Following recent experimental results at sea (N. P. Chotiros, *Proc. Oceans '89*) that suggest the existence of a previously undetected type of acoustic wave in sandy sediments, an experiment was designed to detect and measure the speed of acoustic waves in an isolated environment. The experiment was conducted in a laboratory tank containing 1 m of unwashed river sediment under a 3-m water column. Observations were made of the travel time and attenuation of a pulse from an acoustic source located above the water-sediment interface to a set of probes below the interface. It was observed that, at normal incidence the pulse traveled at about 1750 m/s, while at shallow grazing angles, the pulse traveled through the sediment at close to 1200 m/s. An interesting possible explanation exists in the Biot model [M. A. Biot, *J. Acoust. Soc. Am.* **28**, 168–191 (1956)], which predicts a slow acoustic wave in porous materials. [Work funded by ONR under NOARL management.]

3:30

7UW8. VLF bottom loss measurements in the Blake-Bahama basin. Jeffrey L. Becklehimer, Stephen O'Hara, and L. Dale Bibee (Naval Oceanographic and Atmospheric Res. Lab., Code 362, Stennis Space Ctr., MS 39529)

Broadband (8–25 Hz) measurements of ocean-bottom reflection loss were made using airgun sound sources and a vertical hydrophone array for grazing angles between 5 and 55 deg. Frequency-wave-number analysis was used to partition the energy received by the hydrophone array into the energy spectrum due to the direct water path and that from the path with one bottom reflection. Bottom loss estimates were then calculated as the ratio of the energy spectrum from these two propagation paths. Comparisons of theoretical and measured

reflection loss are made. An increase in bottom loss is observed near 35 deg and can be explained as a critical angle effect. [Work supported by Office of Naval Technology.]

3:45

7UW9. Attenuation of low-frequency sound in the Black Sea. David G. Browning (New London Lab., Naval Underwater Syst. Ctr., New London, CT 06320) and Robert H. Mellen (Univ. of Connecticut, Groton, CT 06340)

Of the three chemical absorption mechanisms in seawater, only two (magnesium sulphate and boric acid) are dominant in most ocean areas. However, in some reduced salinity areas such as the Baltic Sea it has been found that the third mechanism (magnesium carbonate) can make a very significant contribution. An analysis is made of the chemical constituents of the Black Sea, which has a salinity that is approximately one-half that of standard seawater. The resulting values of low-frequency attenuation are computed utilizing the global attenuation model. These results are compared to values in the Baltic Sea and other ocean areas.

4:00

7UW10. Radiated noise characteristics of M/V OVERSEAS HARRIETTE, a modern cargo ship. Paul T. Arveson and David J. Vendittis (David Taylor Res. Ctr., Code 1945, Bethesda, MD 20084)

Low-frequency ambient noise in the ocean is often influenced by radiated noise from merchant shipping. In an effort to quantify this noise, a series of extensive and carefully planned measurements were made of the radiated noise of M/V OVERSEAS HARRIETTE, a 25 525-deadweight ton 567-ft cargo ship powered by a direct-drive low-speed diesel engine. This ship and its power plant are typical of many modern merchant ships. The radiated noise data show high-level tonal frequencies from the ship's service diesel generator, main engine firing rate, and blade rate harmonics due to propeller cavitation. Directivity measurements taken at many angles under the ship indicate that the radiation is generally dipole in form at lower frequencies, as expected. There are some departures from this pattern that may indicate hull interactions. Blade rate fundamental levels show good agreement with predicted levels. [Work supported by NRL.]

4:15-4:30

Break

4:30-5:30

Panel Discussion

THURSDAY AFTERNOON, 2 MAY 1991

SCHAEFER, 1:30 P.M.

Meeting of Accredited Standards Committee S3 on Bioacoustics

to be held jointly with the

U.S. Technical Advisory Group (TAG) Meetings for ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock

L. A. Wilber, Chair S3
422 Skokie Boulevard, Wilmette, Illinois 60091

H. E. von Gierke, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43
and ISO/TC 108/SC4
1325 Meadow Lane, Yellow Springs, Ohio 45387

V. Nedzelnitsky, U.S. Technical Advisor for IEC/TC 29
National Institute of Standards and Technology, Bldg. 233, Rm. A149, Gaithersburg, Maryland 20899

Standards Committee S3 on Bioacoustics. The current status of standards under preparation will be discussed. In addition to those topics of interest, including hearing conservation, noise dosimeters, hearing aids, etc., consideration will be given to new standards which might be needed over the next few years. Open discussion of committee reports is encouraged.

The international activities in ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock, will be discussed. The chairs of the U.S. TAGs for ISO/TC 43 (H. E. von Gierke), IEC/TC 29 (V. Nedzelnitsky), and ISO/TC 108/SC4 (H. E. von Gierke), will report on current activities of these Technical Committees and Subcommittees. Plans for the next meetings of ISO/TC 43 (in Australia in December 1991), and for IEC/TC 29 (in New Zealand, from 25–29 November 1991) will be discussed.

THURSDAY AFTERNOON, 2 MAY 1991

LINCOLN, 3:00 P.M.

Meeting of Accredited Standards Committee S1 on Acoustics

to be held jointly with the

**U.S. Technical Advisory Group Meetings for ISO/TC 43 Acoustics
and IEC/TC 29 Electroacoustics**

G. S. K. Wong, Chair S1

Institute for National Measurement Standards, National Research Council, Ottawa, Ontario K1A 0R6, Canada

H. E. von Gierke, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43
1325 Meadow Lane, Yellow Springs, Ohio 45387

V. Nedzelnitsky, U.S. Technical Advisor for IEC/TC 29

National Institute of Standards and Technology, Building 233, Rm. A149, Gaithersburg, Maryland 20899

Standards Committee S1 on Acoustics. Working group chairs will report on their progress in the preparation of standards on methods of measurement and testing, and terminology, in physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound. Work in progress includes measurement of noise sources, noise dosimeters, integrating sound-level meters, and revision and extension of sound-level meter specifications. Open discussion of committee reports is encouraged.

The international activities in ISO/TC 43 Acoustics, and IEC/TC 29 Electroacoustics, will also be discussed. The Chairs of the respective U.S. Technical Advisory Groups for ISO/TC 43 (H. E. von Gierke), and IEC/TC 29 (V. Nedzelnitsky), will report on current activities of these Technical Committees. Plans for the next meetings of ISO/TC43 (in Australia in December 1991), and for IEC/TC 29 (in New Zealand, from 25–29 November 1991) will be discussed.

Session 8EA

Engineering Acoustics and Underwater Acoustics: Memorial Session for Laurence Batchelder

Stanley L. Ehrlich, Chair

Raytheon Company, Submarine Signal Division, 1847 West Main Road, Portsmouth, Rhode Island 02871-1087

Chair's Introduction—8:30

Invited Papers

8:35

8EA1. Laurence Batchelder: Inventor and patent reviewer. Daniel W. Martin (7349 Clough Pike, Cincinnati, OH 45244)

An inventor for 23 U.S. patents, assigned to Submarine Signal Company and Raytheon Company, Batchelder contributed numerous inventive concepts to underwater sound transducers and arrays, and to systems for hunting, detection, and ranging. For 41 years he regularly reviewed for the *Journal* a total of 3035 patents on transducers, systems, and measuring devices, mostly in underwater sound, and contributed his last review only a month before his death. Examples from his patents and from his insightful reviews will be shown.

9:00

8EA2. Sound intensity in the transducer near field. J. Tichy (Graduate Prog. in Acoust., Appl. Sci. Bldg., Penn State Univ., University Park, PA 16802)

One of the principal research domains of Larry Batchelder was transducers. This paper intends to contribute to this memorial session by summarizing the use of intensity technique for the analysis of sound radiation from transducers. Examples of near-field measurements will reveal how intensity technique can identify both the radiation of power and the acoustical coupling between various surface elements.

9:25

8EA3. Recollections of a contemporary of Laurence Batchelder, concerning "Preferred reference quantities for acoustic levels, ISO 1683-1983." Robert W. Young (Consultant in Acoustics, 1696 Los Altos Rd., San Diego, CA 92109)

Laurence Batchelder became a member of the Acoustical Society of America in 1946, having already been employed by the Submarine Signal Company for 17 years in the development of underwater Sound Navigation and Ranging equipment. He was a strong and successful proponent of referring an acoustical level to one unit of the *Système International* as illustrated in the following. In 1946, sound-pressure level in water at a stated distance from a SONAR projector was typically referred to 1 dyn/cm²; sound-pressure level radiated by a submarine was typically referred to 0.0002 dyn/cm²; sound-pressure level in air measured by a standard sound-level meter was referred to 0.0002 dyn/cm². At a request of the U. S. Navy in 1961, American Standards Association Working Group S1-44 was appointed to propose reference quantities for acoustical levels. After much negotiation, S1. 8-1969 was approved, including a new reference pressure 10^{-5} dyn/cm² = 1 micropascal (μ Pa) for liquids; 2×10^{-4} dyn/cm² = 20 μ Pa was retained as the reference for gases. Also in 1961, when the British Standards Institution held the Secretariat for Technical Committee 43, Acoustics, of the International Organization for Standardization (ISO), the Netherlands Member Body proposed that the single reference pressure for sound-pressure level in all gases and fluids be 2×10^{-5} N/m² = 20 μ Pa. After a 12-year succession of drafts and ballots, during which the BSI relinquished the Secretariat for TC43 to Denmark, a new Working Group 43-14 was appointed whose effort culminated in ISO 1683-1983; 20 μ Pa was confirmed as the reference pressure for air, and 1 μ Pa was standardized for media other than air. The goal proposed by The Netherlands in 1961—only one reference pressure for sound-pressure levels in all gases and fluids—has not been attained.

8EA4. Laurence Batchelder's contributions to national and international standards. William W. Lang (IBM Acoust. Lab., C18B/704, P.O. Box 950, Poughkeepsie, NY 12602)

Over a period that spanned more than 4 decades, Laurence Batchelder made significant contributions to the development of both national and international standards. With patience, good humor, and an untiring willingness to see the work completed, he participated as committee member and chairman in the preparation of many of the most important standards. His standards work started in the early days of the ASA's only standards committee, Z24, and continued through the ASA sectional committee S1 (acoustics), S2 (shock and vibration), S3 (bioacoustics), S12 (noise), to the chairmanship of ASACOS (ASA committee on standards). For many years, he chaired the ASME's Y10.15 committee on letter symbols for acoustics. In the international arena, he served lengthy tenures working for TC43 (acoustics) of the International Organization for Standardization (ISO) and as chairman of TC29 (electroacoustics) of the International Electrotechnical Commission (IEC). He headed U.S. delegations to plenary meetings of ISO/TC43 and IEC/TC29 for more than a decade. His influence on the development of both national and international standards was profound.

10:15

8EA5. Laurence Batchelder's contributions to engineering acoustics and underwater acoustics. Leon R. Warren and Kenneth D. Rolt^{a1} (Submarine Signal Div., Raytheon Co., 1847 W. Main Rd., Portsmouth, RI 02871)

During Laurence Batchelder's career, while underwater acoustics was still in the early stages of development, he was associated with several of the outstanding pioneers in the field, including Professor G. W. Pierce, Dr. H. C. Hayes, and H. J. W. Fay. During his 40 years as an engineer with Submarine Signal Company and its successor, the Submarine Signal Division of Raytheon Company, his technical contributions were significant and embraced a number of engineering disciplines. The challenging technical environment of the 1930s and 1940s and the small group of Government and industry engineers involved with the development and applications of underwater sound equipment produced a working atmosphere ideally suited to his analytical and creative talents. His contributions included developments in the JL series of listening equipment, the QB and QC series of echo ranging equipment, Fathometers[®] and echo sounding equipment, and participation in the 1937 Guantanamo Bay Expedition, a joint effort involving scientists from the Navy and Woods Hole Oceanographic Institution. This expedition established for the first time the close relationship between underwater acoustics and oceanography. His many published works include the post-war report, "Sonar in the German Navy," prepared for the U. S. Navy. ^{a1}Currently on educational leave at MIT Ocean Engineering, 77 Massachusetts Ave., Cambridge, MA 02139.

10:40

8EA6. Laurence Batchelder's contributions to the Acoustical Society of America. Stanley L. Ehrlich (Submarine Signal Div., Raytheon Co., 1847 W. Main Rd., Portsmouth, RI 02871-1087)

Laurence Batchelder was a consistent, well-diversified contributor to the Acoustical Society of America from the time he joined the Society in 1946 throughout the remainder of his life. When you ask different members of the Society what he did, you are likely to receive a unique answer, because he shared his expertise with each of them in a different and very personal way. Some of this interaction is being presented by other speakers in this session. A few chronological highlights of his ASA career are that Larry was elected a Fellow of the Society in 1949, served on the Executive Council from 1952-1954, and again as President-Elect, President, and Past President from 1960-1963, received the first Distinguished Service Citation awarded by the Society in 1972, and most recently was awarded the Gold Medal for 1985, the highest award of the Society for contributions to acoustics.

Contributed Paper

11:05

8EA7. Design of a direct audio communicator for divers. Richard W. Ranlet (Submarine Signal Div., Raytheon Co., Box 360, Portsmouth, RI 02871)

An underwater communicator was developed to evaluate the suit-

ability of direct audio voice transmission as a *practical* means to enable close-range communications between divers using open-circuit SCUBA. A communicator that transmits speech directly into the water through an omnidirectional acoustic projector was designed and human engineered to yield a practical working prototype suitable for normal open-water dive conditions. This system requires no specialized equipment to

enable reception by either submerged swimmers or divers. A series of in-water tests was carried out as an intrinsic part of the design evolution of the device. Once the design was finalized, intelligibility of the communicator was measured by underwater audiological testing using tones, phonetically balanced word lists, and sentence lists. Results indicate average word list intelligibility of 82% correct responses and sen-

tence list intelligibility of nearly 100% (both at 5-m range). Open-water range tests showed comfortable communication range between divers using the device to be greater than 140 ft (43 m) in calm seas. [Work was performed as part of a Master of Science thesis in Ocean Engineering at the University of Rhode Island.]

FRIDAY MORNING, 3 MAY 1991

PRATT A & B, 9:00 TO 11:30 A.M.

Session 8MU

Musical Acoustics and Psychological and Physiological Acoustics: Pitch, Timbre, Time, and Melody

William M. Hartmann, Chair

Department of Physics, Michigan State University, East Lansing, Michigan 48823

Contributed Papers

9:00

8MU1. Perception of vocal pitch vibrato in short tones. Christophe d'Alessandro (LIMSI-CNRS, BP133, 91403 Orsay Cedex, France) and Michele Castellengo (LAM, Univ. Paris IV, 4 Place Jussieu, 75005 Paris, France)

Two different experiments were carried out to examine the perception of short vibrated tones, using synthetic stimuli. A preliminary experiment on long tones showed results identical to those obtained by Shonle and Horan [J. Acoust. Soc. Am. 67, 246-252 (1980)]. The first experiment used a method of adjustment to find the pitch of the vibrated tones, according to their durations (or equivalently, to the number of vibrato cycles). Durations from 80 ms (half-cycle for a vibrato rate of about 6 Hz) to 320 ms (two cycles) were studied. The results indicate that an averaging of F_0 excursion is performed, and that perception may be ambiguous above a threshold of duration, called herein "threshold of fusion." The second experiment used a constant method to estimate the threshold of fusion, defined as the fractional number of cycles where a stimulus is integrated into one tone. Above this threshold, a glissando or two consecutive tones are perceived. The threshold was estimated to be at about the second third of a vibrato cycle beginning at zero phase. It was noticeable that in the first experiment, the subjects were still able to assign a unique pitch to the stimuli above the threshold when the tone is long enough. A concluding discussion points out how these psychoacoustic results may contribute to our understanding of pitch perception for vibrato tones in actual musical performance and explain some aesthetic values governing the production of vibrato by singers.

9:15

8MU2. On the origin of the stretched melodic octave. William Morris Hartmann (Physics Dept., Michigan State Univ., East Lansing, MI 48824)

It is well known that the psychological octave is stretched with respect to the physical octave, i.e., when listeners choose or adjust a melodic octave, the frequency ratio turns out to be greater than 2 to 1. Two explanations have been advanced to explain this effect. Terhardt [J. Acoust. Soc. Am. 55, 1061-1069 (1974)] suggested that the stretched melodic octave is learned from a harmonic octave that is stretched by

partial masking of excitation patterns. The origin of the stretched melodic octave is therefore in the central nervous system. However, Ohgushi [J. Acoust. Soc. Am. 73, 1694-1700 (1983)] argued that the stretched melodic octave may be caused by the refractory delay in primary auditory neurons, as observed in electrophysiological single unit recordings. In his model, pitch is determined by the first several peaks in the interspike interval histogram, which are increasingly delayed for higher frequencies. The origin of the stretched melodic octave is therefore in the peripheral nervous system. The present paper shows that one can choose between central and peripheral models by an octave-judgment experiment that does not involve any pitch information in the auditory periphery. This is done by creating central pitch sensations with the Huggins effect. Data obtained from 2AFC constant-stimuli experiments favor the central-origin explanation of the stretched melodic octave. [Work supported by the NIDCD of the National Institutes of Health.]

9:30

8MU3. Perceptual limits of octave harmony. Laurent Demany, Catherine Semal (Lab. de Psychoacoust., Univ. de Bordeaux 2, 146 rue Léo-Saignat, F-33076 Bordeaux, France), and Robert P. Carlyon (Univ. of Sussex, Brighton BN1 9QG, England)

Three subjects were monaurally presented with dyads of frequency-modulated pure tones approximately 1 octave apart. The tones, with carrier frequencies F_L and F_H , were heard in a background of pink noise and at a low sensation level, so that they were completely resolvable by the subjects' peripheral auditory filters. In experiments 1 and 2, subjects judged on each trial which of two dyads was inharmonic ($F_H \neq 2F_L$); the relative mistuning of the inharmonic dyad $[(F_H - 2F_L)/2F_L]$ was varied independently of F_L and could be either positive or negative; F_L was fixed within trials in one experiment, and varied within trials (from about 8%-20%) in the other experiment. In both experiments, performance monotonically worsened when F_L was increased from 300-2000 Hz; in addition, negative mistunings were better identified as inharmonicities than positive mistunings. In a third experiment, a 41-2AFC procedure was used to assess the detectability of changes in F_H irrespective of their effect on the perception of harmonicity. Performances were not the same function of F_L as in the other two experiments, and were

much better for high F_L values. Thus it is concluded that detection of inharmonicity (experiments 1 and 2) was limited by factors other than those responsible for fine pitch discriminations. These results also have implications with respect to the origin of human listeners' *melodic* octave templates.

9:45

8MU4. Sharpness measurements for musical instrument timbres. Pamela Goad (Systematic Musicology Prog., School of Music, DN-10, Univ. of Washington, Seattle, WA 98195)

The timbres of string and woodwind families were investigated by measuring sharpness and loudness over a 2-octave range and between individual performers. Sharpness is defined as a frequency-dependent weighting of the loudness-critical band rate pattern. It is promising for timbre perception research because it addresses the most salient timbral dimension. Digital recordings were collected with performers 4 m from the microphone in a music rehearsal hall. Performers played diatonic scales at two extreme musical dynamic levels, pianissimo and fortissimo. Across all families, sharpness increases with playing frequencies and with dynamic level, and lies between 0.5–2.0 acums. Loudness variance is larger than sharpness variance. Overlap in sharpness is greatest at the pianissimo levels with the exception of the double reeds. The string family smoothly changes in sharpness across the entire playing range at both musical dynamics levels. The clarinet family has less overlap at the fortissimo level and a slight increase in sharpness for lower pitches. Double reeds exhibit the greatest variance in sharpness. Performer-specific variations are evident, but the general trends in sharpness remain the same.

10:00

8MU5. Normalization process in the human auditory system. Jennifer L. Cho, Michael D. Hall, and Richard E. Pastore (Dept. of Psychol., State Univ. of New York, Binghamton, NY 13901)

Irrelevant variability of source characteristics imposes problems in maintaining perceptual accuracy of an intended message. The perceptual system appears to adjust for differences between sources in a time-consuming manner called normalization which has been investigated with speech stimuli. The present experiment sought to establish auditory normalization with music stimuli using an AX chord discrimination task. Stimuli consisted of four chords produced by five different instruments. Degree of instrument variability was manipulated within subjects by presenting stimuli in three normalization blocks: single instrument, multiple instrument, and multiple-mixed instrument. In these conditions instrument selection remained the same throughout a block of trials, differed across trials, or differed within and across trials. Significant expected differences in response latencies and task accuracy were observed for trials that were composed of different instruments, compared to trials having the same instrument. Results clearly indicate normalization in the perception of music. [Work supported by NSF.]

10:15

8MU6. Measuring similarity of musical timbres. Paul Iverson and Carol L. Krumhansl (Dept. of Psychol., Uris Hall, Cornell University, Ithaca, NY 14853)

Similarity scaling, adjective scaling, and acoustic measurements were used to assess the similarity of 16 digitally recorded tones produced by orchestral instruments playing the same pitch. In similarity scaling experiments, subjects rated the similarity of the tones on a continuous scale. Similarity scaling experiments were run on the complete tones, the first 85 ms of the tones, and on the tones with the first 85 ms removed. In an adjective scaling experiment, subjects rated each complete tone on 21 different adjective scales. Acoustic measurements included correlations between the amplitude spectra of the tones, and

correlations between amplitude spectra at regular intervals within each tone. The results of these experiments were analyzed using correlations, multidimensional scaling, factor analyses, and linear regressions. The ratings from the three similarity scaling experiments were highly correlated with each other, and they were each moderately correlated with the results of the adjective scaling experiment. Acoustic attributes were identified that correlated with the results from the other two methods, thus isolating factors common to all three methods. [Work supported by Sigma Xi, The Scientific Research Society.]

10:30

8MU7. Hysteresis in the perception of directional pitch change. Betty Tuller and Janice Giangrande (Ctr. for Complex Systems, Florida Atlantic Univ., Boca Raton, FL 33431)

It has long been known that pairs of complex tones that are clearly different in pitch may be ambiguous as to the direction of the difference [e.g., R. N. Shepard, *J. Acoust. Soc. Am.* **36**, 2346–2353 (1964)]. The present experiment explores whether the ambiguity in pitch change perception results from the dynamic nature of the perceptual states. Twelve stimulus tones were generated, each 120 ms in duration, and each composed of 10 sequential, harmonically related sinusoidal components. The 12 tones differed by a chromatic step change of the 10 components. Tone pairs consisted of a standard complex tone (D#) followed by one of the other 11 tones. Three conditions were used: (1) randomized presentation of the tone pairs; (2) tone pairs in which the comparison tone increased by one chromatic step in successive tone pairs (1 vs 2; 1 vs 3; etc.); and (3) tone pairs in which the comparison tone decreased by one chromatic step in successive tone pairs (1 vs 12; 1 vs 11; etc.) Consistent with previous reports, when the tone comparisons were randomized the direction of perceived pitch change was ambiguous as the frequency change neared the half-octave. Signature properties of non-linear dynamical systems (hysteresis effects and bistability) were observed in conditions 2 and 3, for all subjects. Specifically, the region of ambiguity shifted markedly as a function of the ordering of tonal comparisons. Implications for geometric models of pitch perception will be discussed. [Work supported by NIDCD and NIMH.]

10:45

8MU8. Perception of small rhythmic variations. Peter Marvit (Dept. of Psychol., Univ. of California at Berkeley, Berkeley, CA 94720)

A study to determine the mechanisms and parameters of the perception of small timing differences in short rhythmic sequences was performed using 13 subjects (which yielded 10 valid subjects) in a completely within-subjects forced choice factorial design. Manipulated variables included three "base" internote intervals (100, 200, 400 ms—corresponding to different tempos) with respective total durations of the rhythmic figures (1200, 2400, 4800 ms), three categories of two patterns each of varying rhythmic complexity from Povel and Essens [*Music Percept.* **2**, 411–440 (1982)], and six variations in internote note onset timings (0–50 ms). Subjects were played pairs of rhythms, one with the timing variation and one without and then vice versa, and were asked to decide whether the rhythms were the same or different. Scores were based on correct discrimination; reaction time was recorded, but varied too much to be useful. All possible combinations of factors were presented to each subject and order of presentation varied for each subject. Results based on differences between interval variations, plus the strong showing of one of the rhythmic patterns, strongly supported several aspects of a model for an internal clock plus a threshold of 30 ms for detection of timing variation. Except for that one pattern, differences between the patterns did not emerge. This casts doubt on the generality of Povel and Essens' complexity determination of rhythmic figures for pure perception (rather than production). Trends of differences in "base" interval (or tempo) and thus the total duration of the rhythmic sequences, although not statistically significant, suggested possible confirmation of the limits of precategorical acoustic storage or a model of rhythmic regularity. The unintentional factor of presentation order

(within a pair of patterns) turned out to be highly significant, further supporting the model of strongly induced internal clocks. Specifically, "regular" rhythms preceding variations produced much stronger discrimination, especially at the 30-ms variation threshold.

11:00

8MU9. Detection of temporal perturbations in a mechanical music performance: Lengthening is more difficult to detect where it makes musical sense. Bruno H. Repp (Haskins Labs., 270 Crown St., New Haven, CT 06511-6695)

Twenty musically literate listeners were repeatedly presented with the first eight bars of a Beethoven minuet, realized with physically regular timing on an electronic piano. On each trial, one of the 47 consecutive eighth-note intervals in the piece was lengthened by a small amount, and the listeners had to mark its location with reference to the score. The resulting detection accuracy profile across the whole musical excerpt showed dips in the places where lengthening would typically occur in an expressive performance. The correlation with the measured timing profile of a particular expert performance (Murray Perahia) was -0.59 ($p < 0.001$). Thus a psychophysical task (detection of temporal irregularity) can engage listeners' performance expectations and reveal the inherent temporal dynamics of substantial music. [Work supported by BRS&G.]

11:15

8MU10. Key-distance effects in melody recognition reexamined. Annie H. Takeuchi and Stewart H. Hulse (Dept. of Psychol., Johns Hopkins Univ., Baltimore, MD 21218)

Key distance is a theoretical measure of the relatedness of two musical keys. Key-distance effects in melody recognition denote that perception of a transposed melody depends on the key distance between the keys of the original and transposed melodies. Key-distance effects in melody recognition are often cited in the psychology of music literature. However, reexamination of a classic study of key-distance effects [J. C. Bartlett and W. J. Dowling, *J. Exp. Psychol.: Hum. Percept. Perform.* 6, 501-515 (1980)] suggests the effects are not robust, and may be an artifact of their experimental design. Three new experiments failed to find the effect Bartlett and Dowling reported, better discrimination of transpositions from nontranspositions in distantly related keys than in closely related keys, even when contextual conditions were designed to enhance the effect. A reversed key-distance effect, better discrimination in closely related keys than in distantly related keys, was found in some conditions. However, the effect was not robust. Different contextual conditions produced different results, suggesting that perception of melodies is influenced by tonal context. [Work supported by NSF.]

FRIDAY MORNING, 3 MAY 1991

INTERNATIONAL E, 8:00 A.M. TO 12:00 NOON

Session 8NS

Noise: Methods for Predicting Noise Outdoors

Gilles A. Daigle, Chair

National Research Council of Canada, Institute for Microstructural Sciences, Ottawa, Ontario K1A 0R6, Canada

Chair's Introduction—8:00

Invited Papers

8:05

8NS1. The prediction of sound-pressure levels outdoors in the Federal Republic of Germany. Alfons Nolle (Federal Environmental Agency, Bismarckplatz 1, D-1000 Berlin 33, Germany)

In the Federal Republic of Germany the VDI Guidelines 2714 and 2720/2 are used for the prediction of sound-pressure levels in the neighborhood of sound sources. These guidelines were developed about 20 years ago and they describe a general frequency-dependent calculation method for sound propagation outdoors under downwind conditions and for different sound sources, mainly industrial plants. Simplified prediction schemes for road and railway traffic have also been derived from the VDI guidelines. The calculation method will be presented. The development of the guidelines have been supported by research work in sound propagation outdoors mainly above soft ground. It will be shown that the calculation methods of the guidelines are able to predict the A-weighted sound-pressure level, which is used in Germany for rating purposes, for the usual situations in the neighborhood of the sound sources. For sound propagation over mainly hard ground, which is a rare case in noise control, nearly no information exists in Germany.

8NS2. A proposed international standard method for calculating the attenuation of sound during propagation outdoors. Joseph E. Piercy (Natl. Res. Council, Ottawa, Ontario K1A 0R6, Canada) and Louis C. Sutherland (Wyle Labs., El Segundo, CA 90245) This method aims to predict noise levels in the community, with engineering precision, from a variety of sources (ideally any source) of known sound power. It represents an integration by the members of working group ISO/TC43/SC1/WG24 of the national methods, principally from The Netherlands, the Scandinavian countries, Germany, and the USA. This method aims to determine the average sound level $L_{Aeq,T}$ under meteorological conditions favorable to propagation (a worst case scenario), and then obtain from that a long-term average sound level $L_{Aeq,LT}$, both as specified in ISO 1996. It consists specifically of octave band algorithms (with center frequencies from 63–8000 Hz) for calculating the attenuation of sound from a point source, or an assembly of point sources.

8NS3. Geometrical ray acoustic model in the case of a linear sound-speed profile for outdoor sound propagation. André L'Espérance (G.A.U.S. Mech. Eng. Dept., University of Sherbrooke, Sherbrooke, Quebec, Canada)

To estimate the effects of temperature, wind, and atmospheric turbulence on the outdoor sound propagation, a heuristic model based on the geometrical acoustic approximation has been developed. This model assumes a constant linear sound-speed gradient, which allows an analytical determination of the direct and reflected curved ray paths and of all the possible additional rays that may appear between a source and a receiver over the ground in presence of a positive sound-speed gradient. The total sound pressure at the receiver is computed by summing the contribution of all the partially coherent rays. In presence of a negative sound-speed gradient, the receiver may be located in the shadow zone. In that case, the sound-pressure level at the receiver is evaluated with the diffraction solution proposed by Berry and Daigle [J. Acoust. Soc. Am. **83**, 2042–2058 (1988)]. Experimental results found in the literature for wind speed gradient and experimental results obtained during summer 1990 for temperature gradient will be presented and compared with the model.

8NS4. Gaussian beam method for computing outdoor acoustic fields. Yannick Gabillet (CSTB 24 rue J. Fourier, 38400 Saint-Martin-d'Hères, France and N.R.C. Inst. of Microstructural Sci., Ottawa, Canada)

Recently, a method using superposition of Gaussian beams has been proposed in the seismological community for the solution of high-frequency wave problems [V. Cervený *et al.*, Geophys. J. R. Astron. Soc. **70**, 109–128 (1982)]. The concept of the theory of Gaussian beams is to solve the wave equation in the neighborhood of the familiar rays by use of the parabolic approximation. The solution associates with the ray a beam with a Gaussian amplitude profile normal to the ray. The approximate global solution for a given source is then constructed by a superposition of Gaussian beams along nearby rays. The method has been applied to typical outdoor sound propagation problems: wind and temperature gradient effect, turbulence effect, diffraction by screen. Comparison with measurements performed in a wind tunnel and with other results of the literature show that the Gaussian beam method is a potentially useful approach.

8NS5. Methods for forecasting impulse noise propagation through the atmosphere. Michael J. White (U.S. Army Construction Eng. Res. Lab., P.O. Box 4005, Champaign, IL 61824-4005)

Predicting the sound levels outdoors at long distances from a given noise source is a challenging and important problem. The propagation depends strongly on several environmental factors, such as the terrain and the wind and temperature profiles. These factors are often beyond our ability to control or measure adequately in a large experiment. In principle, a precise forecast would demand a dense sampling of the environment over time and throughout the media. However, if one were able to obtain a sufficiently dense sampling of the environment, it is likely that the large volume of information gathered would make it of little practical use! In this paper, several models are compared for predicting the sound field at large distances from an acoustic source. Particular attention is given to blast waves in the linear acoustics regime. A new model is proposed for rapid prediction of the mean noise levels. Some discussion is presented on the distributions of measured levels about their mean values and the importance of these distributions in noise prediction and control.

8NS6. Recent progress in outdoor sound propagation prediction by the parabolic equation method. Kenneth E. Gilbert, Xiao Di, and Chulsoo You (Natl. Ctr. for Physical Acoust., University, MS 38677)

During the past 5 years significant progress has been made in predicting long-range sound propagation in the atmosphere using numerical methods based on the parabolic approximation to the Helmholtz equation. The theory for the parabolic approximation is discussed starting from the one-way wave approximation to the two-way (Helmholtz) wave equation. The application of the theory for practical numerical algorithms is briefly reviewed, and some results are presented for sound propagation in a turbulent atmosphere. Finally, a new Green's function approach to the parabolic equation is presented. The new method is approximately 100 times faster than existing parabolic equation methods for outdoor sound propagation. Hence the new algorithm has the potential for useful calculations in the field using a desktop computer. [Work supported by NASA.]

8NS7. Normal mode solution for a downward refracting atmosphere above a complex impedance plane. Richard Raspet and Gordon Baird (Physical Acoust. Res. Group, Univ. of Mississippi, University, MS 38677)

The development of the fast field and parabolic equation solutions to the wave equation has made it possible to solve for the combined effects of refraction in a layered atmosphere and the interaction of sound with a complex impedance ground surface. In many respects the numerical methods have advanced beyond an understanding of the basic phenomena. In an earlier study [J. Acoust. Soc. Am. **89**, 107-114 (1991)], the residue series solution for upward refraction was investigated and provided insight into the nature of the interaction of refraction and ground reflection. In this paper, results of a similar normal mode solution are presented for downward refraction by a bilinear sound velocity profile above a complex impedance ground surface. This model is used to investigate when the surface wave is excited for downward refraction conditions and to develop criteria for the maximum range of cylindrical decay as a function of ground impedance phase and magnitude and the magnitude of the sound velocity gradient. [Work supported by NASA Langley Research Center.]

Contributed Papers

8NS8. A simple exact solution for a point source above a reacting surface. Xiao Di and Kenneth E. Gilbert (Natl. Ctr. for Physical Acoust., University, MS 38677)

An exact analysis is developed for a spherical source in air above a reacting surface. The solution, which is valid for both local and extended reaction, proceeds along the standard route of an expansion of the spherical wave as a superposition of plane waves. Use of a Laplace transform on the plane-wave reflection coefficient leads to a final form that requires a simple numerical integration of an image integral. The resulting solution is exact for all angles of propagation. For a locally reacting ground surface, the image integral is rapidly convergent. A number of limiting cases are investigated and compared with previous solutions. [Work supported by NASA.]

8NS9. Correlation between excess attenuation and geometric or meteorological parameters. Charles T. Moritz and John S. Lamancusa (Ctr. for Acoust. and Vib., Penn State Univ., University Park, PA 16802)

Extensive experimental measurements of sound propagation and vertical sound-speed profiles over level agricultural terrain are presented. A high-power speaker generating pure tones at approximately one-third octave band center frequencies from 160-4000 Hz was used as the noise source. Sound levels were measured and recorded at ranges of 2, 125, 250, and 500 m from the source during various time periods of interest from 3-7 July 1990. Simultaneously, meteorological conditions were measured using a variety of tower mounted anemometers and temperature sensors. A total of 50 sets of 10-min averaged data at each

of 12 frequencies and 4 ranges was obtained. Temperature and velocity profiles were reconstructed using Monin-Obukhov scaling theory. A statistical regression analysis to determine the correlation between excess attenuation and measurable geometric and meteorological parameters was completed. Sample measurements are presented and comparisons are made to the Daigle model [J. Acoust. Soc. Am. **65**, 45-49 (1979)].

8NS10. The use of a one-parameter model to characterize the acoustical properties of asphalt. K. Van Wyk, J. S. Bolton, and P. J. Sherman (Ray W. Herrick Labs., School of Mech. Eng., Purdue University, West Lafayette, IN 47907)

Following a suggestion by Delany and Bazley [J. Sound Vib. **16**, 315-322 (1971)], a single parameter (flow resistivity) fibrous material model has frequently been used to model the acoustical properties of outdoor surfaces. Embleton *et al.* [J. Acoust. Soc. Am. **74**, 1239-1244 (1983)] have shown that an effective flow resistivity for an outdoor surface may be inferred by matching measured and predicted excess attenuations. In this paper, our experience in using such an approach to characterize the acoustical properties of asphalt surfaces will be described. A point source placed directly on the asphalt surface has been used to generate a broadband noise signal. A microphone placed directly on the surface was used to measure 1/3-octave band sound levels as a function of frequency (1-20 kHz) and distance from the source (0.1-12.8 m). The flow resistivity of the surface was estimated by comparing theoretical and measured level differences between 0.1 m and the other measurement locations. It was found that it is possible to obtain consistent estimates of flow resistivity if the data subset (i.e., the range of frequencies and distances to which the theory is compared) is chosen

carefully. It will be demonstrated, for example, that sealed and unsealed asphalt surfaces can be distinguished through estimates of their flow resistivity. However, it has also been found that the best estimate of the flow resistivity of a particular surface tends to vary systematically with the range of frequencies and distances used in the fitting procedure. Thus it appears that a single parameter model is not sufficient to characterize the acoustical properties of an asphalt surface with uniform accuracy over a broad range of frequencies and distances.

11:45

8NS11. Traffic noise prediction from motorways and fast roads. M. Garai, A. Cocchi, and P. Fausti (Istituto di Fisica Tecnica, Università di Bologna, Viale Risorgimento 2, 40136 Bologna, Italy)

A generalized prediction method is presented that has a flexibility that meets the complexity of the Italian laws. The method relies on a computer program that, via an interface with a database, can analyze several measurements taken in the land around a road and compare them to the theoretically predicted noise levels (an improved version of the RLS-81 algorithm is used); the best correlation between the two sets of values can then be enclosed in the prediction algorithm. Thus the program can adapt itself to those site-dependent characteristics that no model can account for. The program can then be utilized to perform all the usual computations, including barrier insertion loss estimates, and to generate noise contour maps. Details will be given about the study of a critical by-pass/motorway surrounding the town of Bologna; this study permitted a thorough check of the program, which was found to be reliable and very easy to use.

FRIDAY MORNING, 3 MAY 1991

CARROLL, 8:30 TO 11:30 A.M.

Session 8PA

Physical Acoustics: Scattering II

Murray Korman, Chair

Physics Department, U.S. Naval Academy, Annapolis, Maryland 21403

Contributed Papers

8:30

8PA1. Scattering of a plane wave at a plane boundary presenting an impedance discontinuity. Jacqueline Naze Tjøtta (Dept. of Mathematics, The Univ. of Bergen, 5007 Bergen, Norway, and School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

The scattering of a sound wave by a boundary presenting an impedance discontinuity has been the object of many theoretical investigations based on the Wiener-Hopf method. Although the method provides an exact solution, the expressions obtained are involved and not very useful in the vicinity of the boundary. The present work addresses the simple case of a plane wave incident on a plane boundary that is made up of two half-planes. One of the half-planes is perfectly rigid, the other is pressure release. The boundary between the two half-planes is a straight line, and the medium over the boundary is a lossless fluid. It is shown that a simple analytical solution can be obtained by taking advantage of the symmetry properties of the problem. The solution is expressed in terms of one-dimensional integrals that can be evaluated numerically. Numerical results for the sound pressure and velocity are presented. The case where the incident plane wave is replaced by a sound beam is briefly discussed. [Work supported by ONR and The Norwegian Research Council for Science and Humanities (NAVF).]

8:45

8PA2. Direct observations of edge-diffracted waves from acoustically soft underwater panels. Jean C. Piquette (Naval Res. Lab., Underwater Sound Reference Detachment, P.O. Box 568337, Orlando, FL 32856-8337)

Sample panels of acoustical materials usually must be fabricated with a sufficiently small size to be accommodated by existing test facil-

ities. However, as test frequency is lowered, the contaminating influence of the diffracted waves originating at the sample's edges becomes increasingly severe. Two techniques for directly observing the edge-diffracted waves when the sample is acoustically soft are described. One method is based on a comparison of measurements obtained from a relatively large sample (having 244×244 cm, or 8×8 ft, lateral dimensions) to measurements obtained from a relatively small sample (having 76×76 cm, or 2.5×2.5 ft, lateral dimensions). The second method involves direct observation of the edge-diffracted wave from an "airbox" sample. This latter sample is literally a "box of air," fabricated using thin plastic walls, and having a geometry identical to the 76×76 -cm material samples used in the comparison-method tests. The frequency range of interest is 1-10 kHz.

9:00

8PA3. Scattering of Lamb waves at the end section of a plate embedded in water. P. Pareige, F. Luppe, and J. Ripoché (Lab. d'Acoust. Ultrason. et d'Electron., U.R.A. C.N.R.S. 1373, Univ. du Havre, Pl. R. Schuman, 76610 Le Havre, France)

The guided nature of a Lamb wave propagating on a water-immersed plate is well known: It is possible to observe its reemission far away from its generation area. Interest is in the phenomena occurring when such a guided wave meets the end of the plate. This study shall not deal with the reflected part of the Lamb wave, but with the forward-scattered wave in water. The experimental study is conducted for different symmetric and antisymmetric incident Lamb waves propagating on an aluminum plate, the end section of which is a right angle. The receiving transducer lies in the plane perpendicular to the diffracting end section. The scattered amplitude is plotted versus the angle between the incidence direction and the diffracted one in that plane. The angular diffraction patterns of the symmetric Lamb waves exhibit a maximum in

the incidence direction, while the ones of the antisymmetric Lamb waves exhibit a minimum in that same direction. A simple model, based on the Huygens' principle, and involving the polarization state of the incident wave, provides results similar to the experimental ones. First results are also presented and discussed for end sections of various angles.

9:15

8PA4. Derivatives of the phase change on reflection from plates: A new analysis of resonance. O. Lenoir, J. L. Izbicki (Lab. d'Acoust. Ultrason. et d'Electron., U.R.A. C.N.R.S. 1373, Univ. du Havre, Pl. R. Schuman, 76610 Le Havre, France), and J. M. Conoir (Le Centre Thomson d'Applications Radars, 78143 Velizy-Villacoublay, France)

Recent experimental works have yielded a renewal of interest on the study of the phase of the scattered pressure by elastic bodies. Recent results, pointed out by Conoir about cylindrical shells, have led the authors to be particularly interested in the derivatives of the phase of the reflection coefficient of an elastic plate, with regard to the longitudinal, shear, and outer liquid velocities, as well as to the frequency. Three theoretical studies have been carried out: The first one consists in calculating the whole derivatives of the phase versus the frequency; the second one in calculating the derivatives versus the parameter chosen to derive the phase; and the third one in calculating the derivative, with respect to the frequency, versus the velocities. The goal is to exhibit the influence of these parameters on the phase when a Lamb mode is excited in an elastic plate. Indeed, the different curves obtained give information about the prevailing character, longitudinal or shear, of the waves related to Lamb modes. They also determine whether a mode is well coupled or poorly coupled with the external medium.

9:30

8PA5. Scattering from slender bodies at high frequency. Michel Tran Van Nhieu (Thomson Sintra ASM, 94117 Arcueil, France)

A scattered field from rigid slender bodies at high frequency is determined by applying the Kirchhoff and slender-body approximations; this last one is the zeroth-order term of the asymptotic expansion of the solution of the problem in the slenderness ratio ϵ [M. Tran Van Nhieu, J. Acoust. Soc. Am. **85**, 1834-1840 (1989)]. It is shown that the pressure scattered from complex targets can be computed from a simple integral and the geometrical knowledge of the transversal cross sections of the body along its longitudinal axis. An analytical expression for the far-field pressure is obtained explicitly. Some numerical results for the monostatic and bistatic target strengths of bodies of various shapes are presented to illustrate the theory.

9:45

8PA6. Signatures from pulse signals scattered at oblique incidence from elongated elastic solids at carrier frequencies in the region of bending resonances. M. F. Werby and Jacob George (NOARL, Code 221, Stennis Space Ctr., MS 39529)

It has been determined that time-domain signals can manifest characteristic signals when the midpulse frequency is in some resonance region. This will be discussed in an upcoming book on resonance scattering theory by Überall. This work is a further extension of that study to illustrate unique signals characteristic of pulse scattering at frequencies associated with bending or flexural resonances. In the earlier work, beat patterns and damped sinusoidal patterns (as a function of time) were associated with single or specific clusters of resonances. Here, alternating damped sinusoidal patterns are associated with intervals of interference patterns that are shown to be associated with bending modes excited at oblique incident angles.

10:00

8PA7. Resonances of Scholte-Stoney waves propagating on cylindrical shells at oblique incidence. P. Remert, O. Lenoir, J. L. Izbicki, and J. Ripoche (Lab. d'Acoust. Ultrason. et d'Electron., U.R.A. C.N.R.S. 1373, Univ. du Havre, Pl. R. Schuman, 76610 Le Havre, France)

This paper deals with the oblique scattering of an incident monochromatic plane wave by a circular cylindrical shell immersed in water. The propagation direction of the incident wave makes an angle α with a normal to the longitudinal axis of the shell. Experiments are carried out over the broad reduced frequency range $0 < ka < 35$, involving the method of isolation of resonances, for two targets: (1) an air-filled aluminum shell with $b/a = 0.94$ (b = inner radius of the shell; a = outer radius of the shell) whose length is infinite with regard to the incident acoustic beam cross section; and (2) the same shell of the finite length: In this case, the incident beam insonifies the entire target. With α varying from 0° to 35° , the backscattered spectra—obtained from the signals scattered by both targets—exhibit wide minima which are due to the reemission of Scholte-Stoney waves. Resonance peaks are located at the same frequencies on the corresponding resonance spectra. The frequency position of these peaks slowly increases with the increasing values of α . Follow-up of these frequencies is easy, especially beyond the shear critical angle, which confirms that these resonances are related to external waves. These new experimental results are similar to theoretical ones obtained for an infinite pipe by Kaplunov *et al.*

10:15

8PA8. Acoustic fields on the surface of a submerged finite cylindrical shell irradiated by plane waves. Xiaoqi Bao, Vasundara V. Varadan, Vijay K. Varadan (Dept. of Eng. Sci. and Mech. Res. Ctr. for the Eng. of Electron. and Acoust. Mater., Penn State Univ., 149 Hammond Bldg., University Park, PA 16802), and C. Audoly (GERDSM, Le Brus, Six-Fours, France)

The prediction of acoustic pressure on the surface of baffles is interesting for designers of transducer arrays. A computer simulation program based on a finite/boundary element approach is developed for this purpose. In the simulation, the elastic baffle is modeled by finite elements and the infinite fluid surround the baffle by boundary elements. The two representations are coupled on the surface through eigenmode expansion. The composite program is able to predict the surface acoustic pressure of axisymmetric elastic baffles. The sensitivity and directivity of hydrophone or hydrophone arrays that are mounted on the baffle can then be obtained. A particular example of a baffle that is a cylindrical steel shell with truncated ends is computed. The convergence of the computation is discussed. The results are compared with experimental measurements.

10:30

8PA9. Acoustic scattering from finite elastic cylinders. Gerard Maze, Jean Ripoche (LAUE-URA, Univ. of Le Havre, 76610 Le Havre, France), Xiao-Ling Bao, and Herbert Überall (Catholic Univ., Washington, DC 20064)

Tank experiments on acoustic bistatic and backscattering from elastic cylinders with hemispherical endcaps have been performed at the University of Le Havre and at Catholic University. The resonances apparent in these experiments have been interpreted on the basis of the phase-matching principle [H. Überall *et al.*, J. Acoust. Soc. Am. **61**, 771 (1977)] for surface waves that encircle these objects along closed geodesics. Upon axial incidence, the resonances of meridionally propagating surface waves are visible. Upon axial incidence, the resonances of meridionally propagating surface waves are visible. Upon broadside incidence, circumferential wave resonances are visible as well as the meridional ones. The frequencies of both types of resonances are explained very well by the phase-matching principle. [Work supported in part by the David Taylor Res. Ctr.]

8PA10. Variational scattering calculations for a prolate spheroid. D. E. Freund and R. A. Farrel (Johns Hopkins Univ. Appl. Phys. Lab., Johns Hopkins Rd., Laurel, MD 20723)

Many theoretical methods for computing the scattering amplitude, T , employ an initial approximation (trial field) to the exact fields within, or on the surface of, the scatterers. Variational principles are unique in that they possess the important feature known as quadratic convergence. That is, first-order errors in the trial fields manifest themselves as only second- and higher-order errors in the desired quantity T . Results that were remarkably accurate at all wavelengths were achieved previously [D. E. Freund and R. A. Farrell, *J. Acoust. Soc. Am.* **87**, 1847-1860 (1990)] by using the Schwinger variational principle in conjunction with simple, physically motivated, trial fields for the case of scattering from a soft sphere. Here, the efficacy of these simple trial fields is investigated further by applying them to the more severe test of scattering from an acoustically soft prolate spheroid with a plane wave axially incident. Accurate forward- and backscattering results are found for spheroids with aspect ratios ranging from 2:1 to 100:1. [Work supported by the U. S. Navy under Contract N00039-89-C-0001.]

8PA11. The manifestation of certain classes of resonances as standing waves on elastic shells. M. F. Werby (NOARL, Code 221, Stennis Space Center, MS 39529) and H. Überall (Catholic Univ., Washington, DC 20064)

In an earlier work it was shown that certain classes of resonances excited on elastic solids correspond to standing-wave patterns on the object surface. The observation was in fact implicit in the circumferential nature of such resonances made clear by the early theoretical works of Überall *et al.* and the experimental work of R. Goodman *et al.* The

demonstration was made possible by subtracting the rigid background (in partial wave space) using the background proposed by Flax, Überall, and Dragonette for elastic solids. The background for elastic shells introduced in the upcoming book by Überall on resonance scattering now makes it possible to demonstrate that this is also observed for Lamb-type resonances and this fact will be demonstrated for several submerged elastic shells.

8PA12. Opening rate of the transverse cusp diffraction catastrophe in scattering from oblate penetrable spheroids. Cleon E. Dean (NOARL, Stennis Space Center, MS 39529-5004) and Philip L. Marston (Washington State Univ., Pullman, WA 99164-2814)

Sound scattered by an oblate penetrable spheroid should produce a transverse cusp caustic in the region associated with the rainbow in optics (for relative speed of sound $c_{\text{scatt}}/c < 1$). The principal curvatures of the generic local wave front that produces the far-field transverse cusp are examined. This wave front is shown to generate a caustic curve $(U - U_c)^3 = d_\infty V^3$, where U and V are horizontal and vertical scattering angles, and U_c is the cusp point direction. The far-field opening rate d_∞ is calculated for the transverse cusp. It is shown that d_∞ has a simple dependence on the parameters of the generic wave front. Define the aspect ratio $q = D/H$, where H is the height and D is the equatorial width of the penetrable spheroid. Generalized ray tracing is used to relate q to principal curvatures and shape parameters of the outgoing wave front and hence to d_∞ . Measurements of d_∞ in the optically analogous problem appear to support the calculation. As q goes to $q_{14} \approx 1.31$, the critical value for the generation of a hyperbolic umbilic focal section, the predicted d_∞ goes to infinity. The nature of the divergence was numerically investigated as was the rate at which d_∞ vanishes as q approaches other critical values. The analysis suggests benchmarks for testing numerical scattering codes. [Work supported by ONR.]

FRIDAY MORNING, 3 MAY 1991

INTERNATIONAL C, 9:00 TO 10:45 A.M.

Session 8PP

Psychological and Physiological Acoustics: Localization and Binaural Processes

Richard Freyman, Chair

Department of Communication Disorders, University of Massachusetts, Amherst, Massachusetts 01003

Contributed Papers

8PP1. Effects of age on auditory spatial resolution in the horizontal plane. David W. Chandler and D. Wesley Grantham (Bill Wilkerson Ctr. and Vanderbilt Univ. Div. of Hearing and Speech Sci., 1114 19th Ave. South, Nashville, TN 37212)

The purpose of this research was to determine if auditory spatial resolution in the horizontal plane changes with age. Spatial resolution was measured in static conditions, represented by the minimum audible angle (MAA), and in dynamic conditions, represented by the minimum audible movement angle (MAMA). MAAs and MAMAs were measured for adult subjects 60 to 80 years of age with normal (or near normal) hearing thresholds. All measurements were obtained in a dark-

ened anechoic chamber. Thresholds were measured as a function of stimulus bandwidth, center frequency, and listening azimuth, for both MAA and MAMA. Additionally, MAMAs were measured as a function of the velocity of the moving stimulus source. MAAs and MAMAs of the elderly will be compared with previous measurements for young normal adults that were obtained under the same stimulus conditions [D. W. Chandler and D. W. Grantham, *J. Acoust. Soc. Am. Suppl.* **1** **87**, S64 (1990)]. It is hypothesized that auditory spatial resolution in the horizontal plane diminishes with age because of generalized deficits of central nervous system processing. This diminished resolution should be reflected by poorer MAAs and poorer MAMAs among the elderly subjects than among the young, normal-hearing subjects. [Work supported by NIH.]

8PP2. Monaural spatial resolution in the horizontal plane under dynamic and static listening conditions. D. Wesley Grantham and David W. Chandler (Bill Wilkerson Ctr. and Vanderbilt Univ. Div. of Hearing and Speech Sci., 1114 19th Ave. South, Nashville, TN 37212)

Auditory spatial resolution was measured in the horizontal plane for three normal-hearing adult subjects in a darkened anechoic chamber. In all conditions, subjects' right ears were occluded by an EAR foam insert plug plus an external sound-attenuating muff, providing a total of about 40 dB of attenuation. In the dynamic conditions, the minimum audible movement angle (MAMA) was measured—that is, the angular extent a moving target had to traverse to be just discriminable from a stationary target. In the static conditions, the minimum audible angle (MAA) was measured—that is, the minimum angular separation between two sequentially presented stationary targets that was just discriminable from a single stationary target presented twice in succession. In general, MAMAs and MAAs decreased as stimulus bandwidth increased from 0 Hz (pure tone at 3000 Hz) to wideband. MAAs were extremely variable across and within subjects (varying from 10° to 40° of arc). MAMAs for slow-velocity targets (10°/s) were usually lower than MAAs measured for the same signals, a result that contrasts with the results from binaural resolution tasks [D. W. Chandler and D. W. Grantham, *J. Acoust. Soc. Am. Suppl. 1* **87**, S64 (1990)]. These results suggest that specialized mechanisms sensitive to dynamic stimuli may play a role in monaural spatial resolution. [Work supported by NIH.]

9:30

8PP3. Monaural localization, revisited. Frederic Wightman, Doris Kistler, and Marianne Arruda (Dept. of Psychol. and Waisman Ctr., Univ. of Wisconsin, Madison, WI 53705)

There are numerous reports in the psychoacoustical literature that human listeners can localize sound sources reasonably well with one ear. Since interaural difference cues are presumably eliminated in monaural conditions, the so-called "monaural spectral cues" introduced by pinna filtering are assumed to provide the information necessary for accurate localization in monaural conditions. In these experiments, listeners localize wideband noise bursts presented either in free-field (with one ear occluded) or via headphones (with the signal to one phone either attenuated or disconnected). In the headphone conditions, pinna filtering effects are added digitally, such that the waveforms at a listener's eardrum are nearly the same as those produced by free-field sources. The noise spectrum is scrambled from trial to trial to prevent learning. With the noise bursts presented at about 30 dB SL in free-field, the results are consistent with other recent reports and suggest that some ability to localize, especially in the vertical direction, is retained in monaural conditions. However, when the identical stimuli are presented via headphones, there is no indication that sources can be localized monaurally. In other conditions, listeners localize constant spectrum stimuli, free-field stimuli at 70 dB SL, and binaural headphone stimuli with one ear attenuated. Results from these conditions suggest that monaural localization in free-field is most likely mediated by small head movements, *a priori* knowledge of the stimulus spectrum, and acoustical leakage through the ear-occluding devices used to monauralize the listeners. [Work supported by NIH and NASA.]

9:45

8PP4. Influence of onset cues in lateralization. Richard L. Freyman (Dept. of Commun. Disord., Univ. of Massachusetts, Amherst, MA 01003) and Patrick M. Zurek (MIT, Cambridge, MA 02139)

When pairs of clicks delivered binaurally via earphones are repeated to form a click train, interaural cues at stimulus onset can exert a strong influence on the perceived intracranial position of the entire stimulus [Saber and Perrott, *J. Acoust. Soc. Am. Suppl. 1* **86**, S11 (1989)]. This phenomenon has been observed most clearly under conditions in which the interaural time delays in the ongoing click trains—because they

alternate between two values—are ambiguous. For example, when a single click pair with a 500- μ s time lead to the right is followed by a 250-ms train consisting of alternating diotic and left-leading click pairs, the entire stimulus is lateralized to the right. This paper will describe conditions in which onset cues dominate the perceived lateral position as well as those in which steady-state cues determine the lateralization. It will also describe stimulus manipulations that appear to produce a release from the influence of the onset cues. For example, a brief silent gap inserted in the middle of a click train released the listener from the influence of the original onset, while a brief burst of binaural white noise presented during the middle of the click train did not. [Work supported by NSF and NIH.]

10:00

8PP5. Differential lateralization interference for interaural time and interaural level differences. Laurie M. Heller and Virginia M. Richards (Dept. of Psychol., 3815 Walnut St., Univ. of Pennsylvania, Philadelphia, PA 19104)

Lateralization thresholds were obtained in a 2IFC task using 200-ms noise bands presented with either an interaural time or level difference. The elevation in lateralization threshold (interference) caused by a simultaneously presented noise band was measured. When the ITD was applied to a 50-Hz-wide noise band centered at 500 Hz, there was minimal interference from a diotic 400-Hz-wide noise band centered at 4000 Hz. When the ITD was applied to the noise band centered at 4000 Hz, there was substantial interference from a diotic band of noise centered at 500 Hz [D. McFadden and E. G. Pasanen, *J. Acoust. Soc. Am.* **59**, 634–639 (1976)]. In contrast, the interfering effect of a diotic noise band was greater when the ILD was applied to the 500-Hz band of noise than when it was applied to the 4000-Hz band of noise. A similar pattern of results was obtained when the ITD or ILD of the interfering band of noise was randomized from interval to interval, although randomization produced greater interference in all conditions. The complementary pattern of results for time and level differences favors an explanation based on averaging between simultaneous interaural parameters rather than a masking-based explanation.

10:15

8PP6. Modeling of binaural discrimination of multiple sound sources: A contribution to the development of a "cocktail party processor." H. Slatky (Lehrstuhl fuer allgemeine Elektrotech. und Akust., Ruhr-Univ. Bochum, D-4630 Bochum 1, Germany)

The human hearing system when being stimulated simultaneously by narrow-banded signals from two loudspeakers at different locations is able to determine the directions of both sound sources. However, conventional models of binaural interaction which are based on interaural cross correlation in critical bands fail in discriminating the source directions. The reason is that the cross-correlation functions at the output of the model display severe fluctuations in time. A revised binaural cross-correlation model that contains an additional "recomputation mechanism" will be presented. This mechanism estimates directions and energies of sound sources from cross-correlation functions fluctuating in time. Using this mechanism the results of relevant auditory experiments can be reproduced. This model has been applied to construct a cocktail party processor, which is able to enhance the signal-to-noise ratio considerably.

10:30

8PP7. Changing echo thresholds. Richard Freyman (Commun. Disord., Univ. of Massachusetts, Amherst, MA 01003), Rachel Clifton, and Ruth Litovsky (Univ. of Massachusetts, Amherst, MA 01003)

Echo thresholds for a 4-ms white noise burst were measured in an anechoic chamber with the leading loudspeaker located at 45° left and three lagging loudspeakers at 35°, 45°, and 55° right. The listener judged

whether the echo sound came from the loudspeaker at 35° or 55° right by pressing one of two buttons; correct feedback was provided on every trial. This test noise burst was presented in isolation, or preceded by several conditions. Lead only had a train of nine noise bursts from the leading loudspeaker preceding the test noise burst; Lag only had a train from the lagging loudspeaker at 45° right; Precedence effect (PE) had a train from both leading and lagging loudspeakers at 45° left and right.

The PE condition had the highest echo threshold (14.9 ms), followed by the isolated burst (11.2 ms), with the lowest threshold for Lead only (6.8 ms). Increased threshold under the PE condition indicated that repetition of the echo during the train was necessary to drive up echo threshold compared to the isolated burst. The single source train appeared to enhance the test burst echo following the train. [Work supported by NSF.]

FRIDAY MORNING, 3 MAY 1991

INTERNATIONAL B, 8:00 A.M. TO 12:00 NOON

Session 8SP

Speech Communication: Consonant and Vowel Perception

Sally G. Revoile, Chair
CASS/MTB, Gallaudet University, Washington, DC 20002

Contributed Papers

8:00

8SP1. Thresholds for formant-frequency discrimination of vowels in consonantal context. Diane Kewley-Port and Charles S. Watson (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405)

The discrimination thresholds for shifts in formant frequency were shown to be in the range of 1%–2%, in a recent report to this society [Kewley-Port, *J. Acoust. Soc. Am. Suppl.* 1 **87**, S159 (1990)]. Thresholds for F_1 and F_2 obtained from well-trained subjects listening to vowels under minimal stimulus uncertainty were a factor of 3 lower than earlier estimates. The present experiment extends that study to examine the effects of placing a vowel in a consonantal context. The vowel /t/ was synthesized in CVC syllables for the consonants /b/, /d/, /g/, /z/, /m/, and /l/. For $F_1=450$ Hz, the threshold, ΔF , was the same for isolated /t/ as the ΔF averaged over all CVC contexts, about 12 Hz. For $F_2=2300$ Hz, ΔF was significantly larger (45 Hz) for the vowel in the average CVC context than in isolation (25 Hz). Thresholds for individual CVC's were significantly different from the threshold for isolated /t/, in about one-half of the cases examined. These differences are discussed in terms of the extent of the formant transitions and the durations of the steady-state vowel formants. [Research supported by NIH and AFOSR.]

8:15

8SP2. Are articulations integrated in the perception of vowel height? John Kingston (Linguistics Dept., South College, Univ. of Massachusetts, Amherst, MA 01003)

Using the Garner paradigm [W. Garner (1974)], Kingston [Phonetica (in press)] demonstrated that the acoustic effects of differences in velum height (the frequency separation of the nasal pole and zero = nasalization) and rate of vocal fold vibration (fundamental frequency) which covary with tongue height in vowels are integrated perceptually with the acoustic effect of that articulation (first formant frequency), perhaps because they exaggerate the perceptual value of the latter articulation. The failure to separate perceptually the acoustic effects of these three articulations challenges the claim of direct realists [e.g., C. Fowler, *J. Phonet.* **14**, 3–28 (1986)] that articulatory gestures are the objects of speech perception, but in only a limited way, since the

stimuli were brief and simple enough that they may not have allowed listeners to attribute these acoustic effects appropriately to their articulatory sources. Experiments are currently in progress to test whether similar perceptual integration occurs even when other aspects of the stimuli would allow the acoustic effects to be attributed to coarticulation [R. Krakow *et al.*, *J. Acoust. Soc. Am.* **83**, 1146–1158 (1988)], e.g., is nasalization integrated with first formant frequency in nasal as well as oral consonant contexts? Integration will also be tested more rigorously than in the earlier work, using precepts of signal detection theory.

8:30

8SP3. Assessing the role of F_0 in vowel perception via linear logistic modeling. T. M. Nearey (Dept. of Linguistics, Univ. of Alberta, Edmonton, Alberta T6G 2E7, Canada) and J. E. Andruski (Brown Univ., Providence, RI 02912)

Logistic models provide powerful tools in evaluation of stimulus-response relationships in speech perception [T. Nearey, *J. Phonet.* **18**, 347–373 (1990)]. Excellent fits result when such models are applied to vowel perception data. These models allow insight into the possible normalizing role of F_0 in vowel perception. If (as in many current perceptual accounts) the role of F_0 is restricted to an additive formant normalization factor (for some nonlinear transform of the frequency axis), then optimized logistic models using formant frequencies and the fundamental as predictors should show certain specific patterns of correlation among certain estimated parameters across vowel categories. Preliminary results from the analysis of data in these laboratories indicate that this is indeed the case. Furthermore, F_0 normalization appears to occur nearly independently in “head” and “tail” sections of “hybrid” syllables, where, e.g., the ending portion of a syllable from a male speaker is spliced (following a period of silence) to the beginning portion from a female. [Work supported by SSHRC.]

8:45

8SP4. An approach to the classification of American English diphthongs. Michael Gottfried (Central Inst. for the Deaf, 818 S. Euclid Ave., St. Louis, MO 63110)

Six diphthongs of American English (/au, ai, ei, ou, ɔɪ, ju/) were produced by four midwestern American speakers (two male, two female) at two tempos (slow, fast) with differing stress (stressed, unstressed) in two contexts ([b_d], [h_d]). Using a plot of the fundamental frequency and the first three formants derived from linear-prediction-coding (LPC) analysis, the onset and offset of each production was determined. The pattern of formants and fundamental frequency at the onset and offset of diphthongs was used to establish a set of parameters that can classify intended productions of the American English diphthongs in varying stress and tempo conditions with an average accuracy of 93%. Results are also presented for diphthong, target-syllable, and sentence durations. The classification results are discussed with respect to hypotheses concerning the perception of diphthongs.

9:00

8SP5. Auditory-perceptual interpretation of vowels and diphthongs: A progress report. James D. Miller (Central Inst. for the Deaf, 818 S. Euclid Ave., St. Louis, MO 63110)

The auditory-perceptual interpretation of vowels [Miller, J. Acoust. Soc. Am. 85, 2114-2134 (1989)] will be reviewed briefly and a similar approach to diphthongs, based on spectral glides, will be presented. The relations between acoustic descriptions in the auditory-perceptual space and articulatory descriptions will be outlined. Also, issues relating to the roles of formants versus spectral patterns and the precise meaning of target zones will be presented. The distinction between target zones for vowels produced as steady states as opposed to target zones for vowels produced as spectral glides will be emphasized. Preliminary criteria that may serve to distinguish these will be mentioned. Finally, recent data, which appear to be consistent with the auditory-perceptual approach, will be presented.

9:15

8SP6. "Correction" in the perception of filtered vowels. Elizabeth E. Shriberg (Dept. of Psychol., 3210 Tolman, Univ. of California, Berkeley, Berkeley, CA 94720) and John J. Ohala (Univ. of Alberta, Edmonton, Alberta T6G 2E7, Canada)

Two studies examined the effect of cues to channel characteristics on listeners' perception of low-pass-filtered (1000-Hz) vowels. In experiment 1, 85-ms steady-state portions of 11 English vowels excised from digitized natural speech were preceded by a sentence. Following an unfiltered sentence, filtered front vowels were largely perceived as back vowels; however, following a filtered sentence, the effect was reduced, and front vowels were "corrected" at high rates ($\chi^2 = 108.45$, $p < 0.001$). In experiment 2, the sentence was eliminated and a "masker" was added to the filter-reject region of the stimuli; again, a striking increase in front-to-back confusions occurred when vowels were filtered, and a decrease in these errors occurred when high-frequency

noise was added to the filtered vowels ($\chi^2 = 63.74$, $p < 0.001$). In both experiments, a small but stable increase in back-to-front errors in conditions containing cues to filtering was also observed. Results suggest that in these conditions, but not in those lacking cues, listeners determined which filtered vowels were actual front vowels.

9:30

8SP7. Different discrimination strategies for vowels and consonants. Bert Schouten and Arjan van Hessen (Res. Inst. for Language and Speech, Univ. of Utrecht, Trans 10, 3512 JK Utrecht, The Netherlands)

In speech perception research, one is often interested in the relationship between phoneme identification and discrimination of stimuli drawn from the same stimulus continuum. If discrimination performance is completely predictable from identification, perception is often said to be completely categorical: In both tasks, subjects (can) only use phoneme labels. How should one compare a one-interval identification task with a two-interval forced-choice discrimination task in which subjects have to determine the order of the stimuli? Using standard SDT assumptions about optimal processing [D. M. Green and J. A. Swets, *Signal Detection Theory and Psychophysics* (New York, 1974)], it was found that identification and discrimination of natural vowels were nearly equivalent, but with natural stop consonants identification d' was, paradoxically, twice as high as discrimination d' . It was concluded that subjects probably use a different strategy for the discrimination of natural stops: They do not subtract the traces of the two stimuli, but the estimated distances between the stimuli and the phoneme prototypes. Such an assumption yields d' values that are twice as high as the standard values.

9:45

8SP8. The nonlinear dynamics of categorical perception. Betty Tuller, J. A. Scott Kelso, Pamela Case, and Mingzhou Ding (Ctr. for Complex Syst., Florida Atlantic Univ., Boca Raton, FL 33431)

Much research on speech perception over the years has focused on uncovering examples of the nonlinear relationship between acoustics and perception (so-called "categorical perception"). However, little is known concerning the dynamics of this phenomenon. In a variation of the classical categorical perception paradigm, the present experiment explored gradual increases or decreases in a single acoustic parameter. The resulting patterns of perceptual change showed rich dynamics, including hysteresis, "anticipation," a single boundary, and the progression from hysteresis to anticipation over multiple trials. A dynamical system that could account for these perceptual patterns was investigated by specifying a potential function that corresponds to the layout of phonetic (attractor) states, and how that layout alters as the acoustic parameter changes. The model reproduces the observed features of the experimental data, and makes further predictions about perception, currently being tested. [Work supported by NIDCD and NIMH.]

10:00-10:15

Break

10:15

8SP9. The role of multiple acoustic properties in specifying the internal structure of phonetic categories. Philip Hodgson and Joanne L. Miller (Dept. of Psychol., Northeastern Univ., Boston, MA 02115)

The extent to which a phonetic segment may be specified by more than one acoustic property was investigated using the trading relation between vowel duration and closure duration, two properties known to play a role in specifying the voicing contrast for intervocalic bilabial stops. Specifically asked was whether a change in preceding vowel du-

ration results in a comprehensive remapping between closure duration and phonetic category, or whether trading effects are confined to the category boundary. Two series of disyllables were created ranging from /aba/ through /apa/ to */apa/ (an exaggerated /p/) having initial vowel durations of 153 and 250 ms, respectively. Closure duration in each series varied from 20 to 400 ms. A preliminary experiment revealed a standard trading relation, in that the /b/-/p/ category boundary was located at a longer closure duration for the stimuli with a long, compared to a short, preceding vowel. In the main experiment, listeners were asked to judge each disyllable in each series for the goodness of its consonant as a member of the /p/ category. For both series the /p/ category was perceived as having internal structure, with a limited range of stimuli being judged as the best exemplars. Furthermore, the range of best exemplars for the long vowel series was displaced relative to that for the short vowel series toward longer values of closure duration. These findings indicate that the acoustic properties in question trade against each other not only at the phonetic category boundary but also within the category. This results in a comprehensive remapping of phonetic category structure similar to that observed in earlier research for changes in speaking rate. [Work supported by NIH.]

10:30

8SP10. Influence of a syllable's form on the perceived internal structure of voicing categories. Lydia E. Volaitis and Joanne L. Miller (Dept. of Psychol., Northeastern Univ., Boston, MA 02115)

The role of syllable structure on voice-onset time (VOT) was examined by comparing VOT values in consonant-vowel (CV) and consonant-vowel-consonant (CVC) syllables, across a range of speaking rates. In a production study, when CV and CVC syllables were equated for overall duration, VOT values were found to be consistently shorter for the CVC than the CV syllables. Furthermore, when syllables were equated for CV duration, VOT values for CV and CVC syllables tended to be equal, suggesting that speakers were producing VOT values with regard to the syllable's CV duration. In a subsequent perception study, listeners adjusted for these changes in VOT by altering three aspects of category structure in relation to the syllable's CV duration, and not to its overall duration—the location of the voiced-voiceless category boundary, the upper limit of the voiceless category, and the range of "good" exemplars that lies between those two boundaries. These findings support the notion that listeners perceptually restructure their phonetic categories so as to accommodate changes in VOT that occur in production as a result of the syllable's phonological context, as well as its speaking rate. [Work supported by NIH.]

10:45

8SP11. Independence of scene analysis and the speech module. D. H. Whalen and Alvin M. Liberman (Haskins Labs., 270 Crown St., New Haven, CT 06511)

Even the first example of the duplex effect [Rand, *J. Acoust. Soc. Am.* 55, 678-680 (1974)] gives evidence that speech perception can bring together portions of the speech signal that scene analysis says are separate: Although a formant transition on one ear sounds like a nonspeech "chirp," its speech information is used by the syllable on the other ear. The present work explores the competition between these two organizations of the signal by extending previous work [Whalen and Liberman, *Science* 237, 169-171 (1987)] in which F_3 formant sinusoids provided speech information both below and above the intensity level at which the nonspeech aspect of the tone was just perceptible. As an extension, alternative "scenes" were made plausible for the transitional tones by (1) extending them 50 ms before the syllable; (2) extending them 50 ms after the syllable; (3) creating two harmonics based on the tone; and (4) creating four harmonics based on half the frequency of the tone. Identification of the speech sounds was always above chance, indicating the success of the speech module. However, the harmonic series (below the detection level) and the precursor tone (below and above

the detection level) reduced the accuracy of the speech judgments, suggesting competition with scene analysis. Scene analysis and speech perception, though independent, are often in competition for the same signal, and speech takes precedence in a wide array of circumstances. [Work supported by NIH Grant HD-01994.]

11:00

8SP12. Effects of lexical status on perceptual organization in duplex perception. Lynne C. Nygaard (Speech Res. Lab., Dept. of Psychol. Indiana Univ., Bloomington, IN 47405)

Duplex perception occurs when a synthetic syllable is split so the third-formant transition is presented to one ear and the rest of the syllable (the base) is presented to the other ear. Listeners report hearing two distinct percepts—a complete syllable in the ear with the base and a nonspeech chirp in the ear with the transition. A modification of this duplex phenomenon can be created by presenting a third-formant transition in isolation to one ear and the same transition electronically mixed with the base to the other ear. In this case, the transition information fuses to form a chirp percept in the center of the head and the syllable in the other ear becomes clearer than the one produced with the standard duplex procedure. Nygaard [*J. Acoust. Soc. Am. Suppl.* 1 87, S71 (1990)] found that when the spectral composition or onset frequency of the isolated transition was varied relative to the complete syllable base, both phonetic integration and nonphonetic fusion remained remarkably intact even with large differences in spectral composition between components. In a series of experiments, the lexical status of the syllable base was varied to determine the effect of lexical information on perceptual organization of acoustic components that differ in spectral composition. It was found that the lexical status of the eventual phonetic percept influenced the phonetic integration of acoustic components into syllable percepts, but had no effect on fusion of the third-formant transitions to create a centered chirp percept. These results suggest that lexical information contributes to the perceptual grouping of acoustic components into phonetic percepts.

11:15

8SP13. An investigation of locus equations as a source of relational invariance for stop place categorization. Harvey M. Sussman (Dept. of Linguistics and Speech Commun., Univ. of Texas, Austin, TX 78712)

Locus equations were investigated as a potential higher-order metric capable of illustrating relational invariance for place of articulation in voiced initial stop consonants independently of vowel context. Locus equations are straight-line regression fits to data points formed by plotting onsets of F_2 transitions along the y axis and their corresponding midvowel nuclei along the x axis. Twenty subjects, 10 male and 10 female, produced /b/v/t/, /d/v/t/, and /g/v/t/ tokens for 10 vowel contexts. Each CVC token was repeated in a carrier phrase five times yielding 150 tokens per subject. Formant measures were obtained using the MacSpeech Lab II speech analysis system. Locus equation scatterplots revealed extremely tight clustering of points around the regression line that were consistent across speakers and gender. Derived slope and y -intercept parameters were significantly different across stop place categories. The relative value of $F_{2\text{onset}}$ as it linearly changes in relation to the coarticulatorily produced vowel reflects an acoustic correlate of relational invariance for stop place. A discriminant analysis using $F_{2\text{onset}}$ & vowel as predictors showed 82%, 78%, and 67% classification rates for labial, alveolar, and velar place. Using derived slope and y -intercept values as predictors led to 100% classification into stop place categories. A neurobiologically oriented perspective on the invariance issue is explored and a brain-based recognition algorithm for stop place integrating burst and F_2 cues is offered. [Work supported by NSF.]

8SP14. The role of transition duration in the perception of speech and nonspeech analogs of place of articulation. Renée A. E. Zakia (DMLL, Morrill Hall, Cornell Univ., Ithaca, NY 14853) and Richard N. Aslin (Univ. of Rochester, Rochester, NY 14627)

A previous report [R. Zakia, J. Acoust. Soc. Am. Suppl. 1 87, S117 (1990)] demonstrated that at a formant frequency pattern ambiguous between alveolar and velar, subjects identify synthetic speech stimuli with longer transition durations as velars and stimuli with shorter transition durations as alveolars. These results suggest the operation of either (1) an articulatory property characteristic of velars (longer transition durations distinguish velars because of their slower articulatory release), or (2) differential perceptual sensitivity to transition duration for spectral patterns characteristic of velars. To evaluate these possibilities, nonspeech analogs of a formant pattern ambiguous between an alveolar and a velar were generated with transition durations ranging between 20 and 50 ms in 5-ms steps. In contrast to the previous identification task, the nonspeech stimuli were presented in a same-different discrimination task. For comparison purposes, this same-different task was also conducted with the speech stimuli. Transition duration was an effective cue to discrimination of both speech and nonspeech stimuli, suggesting that the link to articulatory mechanisms is not essential. However, discrimination performance for the speech stimuli was generally poorer than for the nonspeech stimuli, suggesting that the link to articulatory mechanisms is not essential. However, discrimination performance for the speech stimuli was generally poorer than for the nonspeech stimuli, suggesting either that within-category judgments are more difficult to make for broadband signals or that perception of the speech signals as phonetic segments interferes with auditory discrimination.

8SP15. The duration and perception of English epenthetic and underlying stops. Sook-hyang Lee (Dept of Linguistics, Ohio State Univ., Columbus, OH 43210)

In American English, an intrusive stop occurs before the fricative in words such as *tense* and *false*, making them very much like words with underlying stops, such as *tents* and *faults*. Ohala (1975) treats the inserted stop as an artifact of universal physiological or aerodynamic constraints. But this approach cannot account for the fact that South African English speakers do not insert the stop between sonorant and fricative clusters (Fourakis and Port, 1986). Another approach posits a language- or dialect-specific phonological rule which inserts a phonological segment (Zwicky, 1972). Fourakis and Port (1986) argue against this approach on the grounds that in some pairs the intrusive stop is significantly shorter than the underlying one (although the difference is always very small). This paper presents perception data and duration measurements supporting Zwicky's approach. Phrases with intrusive and underlying stops (*intense* and *in tents*, respectively) in citation forms produced by three speakers of midwestern dialects were presented over earphones in random order for subjects to identify. Identification was very poor, just at chance level. Also, duration measurements of the silence gap between the /n/ and /s/ in these words show no significant difference, contrary to Fourakis and Port's findings. Moreover, token judgments in the perception experiment show very poor correlation with the durations except for one speaker, implying that whatever duration differences there are might not be a crucial cue that listeners exploit for labeling the words with epenthetic and underlying stops.

FRIDAY MORNING, 3 MAY 1991

LIBERTY B, 8:25 TO 11:45 A.M.

Session 8UW

Underwater Acoustics: Signal Processing

John Glattetre, Chair

Norwegian Defense Research Establishment, P.O. Box 115, N-3191, Horten, Norway

Chair's Introduction—8:25

Contributed Papers

8:30

8UW1. Uncertain field signal processing. Anthony M. Richardson and Loren W. Nolte (Dept. of Elect. Eng., Duke Univ., Durham, NC 27706)

Classical matched-field processing assumes a known ocean environment and known array sensor positions. It is well known that source localization using this method can be quite sensitive to this knowledge. The uncertain field processor approach is based on signal detection and estimation theory and allows one to incorporate, *a priori*, the uncertainties in the sound-speed profile and/or the array sensor positions. This approach is illustrated for several specific situations for a deep ocean environment. [Work supported by ONR and North Carolina Supercomputing Facility.]

8:45

8UW2. Parallel simulation of broadband ambiguity surfaces. Peter C. Mignerey (Naval Res. Lab., Appl. Ocean Acoust., Washington, DC 20375)

A connection machine was used to implement parallel algorithms for the solution of the normal mode equation and for matched-field processing over a large frequency band. The all-acoustic layered model consists of piecewise linear sound-speed and density profiles with a free surface and an acoustic half-space below the bottom. Individual processors are dedicated to one particular mode with the set of processors spanning the frequency band for numerous sound-speed profiles. Ambiguity surfaces as a function of range, depth, and frequency were generated in parallel using a linear estimator. These images show consid-

erable variation in the sidelobes as a function of frequency while the main peak remains at the source location. [Work supported by the Office of Naval Research.]

9:00

8UW3. Broadband modeling and source localization of a PRN projector tow. Evan K. Westwood (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX 78713-8029)

A broadband ray model is used to simulate data from a deep water experiment (TAGEX 87) in which a pseudo random noise (PRN) source is towed past a 24-element, bottom-moored array. The source spectrum contains a series of evenly spaced lines from 60–100 Hz. Simulated shade plots of received spectrum level versus frequency and time exhibit excellent agreement with the data. A broadband matched field algorithm is used to localize the source in range as a function of time. The algorithm consists of matching the recorded cross spectrum $R_i R_j^*$ between two phones with simulated cross spectra $S_i S_j^*$ at a series of ranges. The correlations between recorded and simulated cross spectra are added coherently as a function of frequency to obtain the localization matrix entry: $L_{ij} = \sum_f [R_i R_j^*] [S_i S_j^*]^*$. Successful localizations of the PRN source are obtained at ranges from 0–30 km. [Work supported by Naval Ocean Systems Center, Code 733, under contract N00039-88-C-0043-1-43.]

9:15

8UW4. A new technique for detecting Gaussian signals in non-Gaussian noise. H. Vincent Poor (Dept. of Elect. Eng., Princeton Univ., Princeton, NJ 08544)

The signal and noise processes arising in passive underwater acoustic detection are usually modeled as being random processes, with the dominant noise often being best modeled as a non-Gaussian process [H. V. Poor and J. B. Thomas, *J. Acoust. Soc. Am.* **63**, 75–80 (1978)] due to the effects of noise phenomena resulting from sources such as cracking ice, marine animals, or surface shipping. A new technique for the detection of Gaussian signals that can be modeled as being produced by linear stochastic systems, in the presence of such non-Gaussian noise, has been developed. This technique is based on an approximation to the likelihood-ratio statistic [H. V. Poor, *An Introduction to Signal Detection and Estimation* (Springer-Verlag, New York, 1988)] for such situations. This likelihood-ratio approximation is in turn based on the *Masreliez approximation* of nonlinear filtering [R. Vijayan and H. V. Poor, *IEEE Trans. Commun.* **38**, 1060–1065 (1990)], in which the predicted state probability density (i.e., the time update) of the underlying stochastic system is approximated with a multivariate Gaussian distribution. This approximation allows calculation of the likelihood-ratio statistic using a pair of sufficient statistics satisfying a simple nonlinear recursion. An approximation to the locally optimum detection statistic [Poor, *op. cit.*] for this situation has also been derived, by considering

the limiting behavior of the approximate likelihood-ratio statistic as the signal-to-noise ratio vanishes. [Work supported by the U.S. Office of Naval Research under Grant N00014-89-J-1321.]

9:30

8UW5. Joint estimation of range and bearing in the presence of fluctuation. James F. Bartram and Sudha S. Reese (Raytheon Co., Submarine Signal Division, 1847 W. Main Rd., Portsmouth, RI 02871)

Active sonar joint range-bearing estimation is treated, based on the use of a signal with a large bandwidth–time (WT) product, specifically one with linear frequency modulation (LFM), together with a processing scheme in which range is estimated by means of pulse compression and time-delay measurement, and bearing is estimated using split-array differential phase processing. The degradation in range resolution caused by fluctuation is derived using a simple channel spreading model, whose parameters are B , the Doppler spread, and L , the delay spread, while the degradation in bearing precision is evaluated on the assumption that the physical fluctuation process will cause the returning signal to become normally distributed. It is shown that temporal segmentation of the waveform can have a beneficial effect. The optimum number of segments to use is determined, depending on the WT product of the signal, on the one hand, and the BL product characteristic of the fluctuation process on the other. The effect of using a different type of frequency modulation in the signal design remains for future study.

9:45

8UW6. Modal decomposition of the pressure field on a vertical line array. David J. Thomson, Gordon R. Ebbeson, and Brian H. Maranda (Defence Res. Establishment Pacific, FMO Victoria, British Columbia V0S 1B0, Canada)

For many applications, it is useful to decompose the acoustic field measured in shallow water into its horizontal wave-number components. Recently, a PE-based method was described for effecting the modal decomposition of the depth-dependent field at a given range in a range-dependent waveguide [D. J. Thomson, *J. Acoust. Soc. Am. Suppl.* **186**, S53 (1989)]. This method should be applicable to the analysis of data obtained with a vertical line array (VLA). However, for practical arrays, the measured field is known at only a limited number of hydrophones, whereas the PE-based decomposition method requires the field to be known at each depth on the computational grid. Therefore, to populate the entire PE grid, it is necessary to interpolate the field between hydrophones and to extrapolate it into the bottom. An interpolation and extrapolation scheme suitable for this purpose is proposed. To illustrate the effectiveness of this reconstruction scheme, modal decomposition is carried out using simulated VLA data generated for an environment representative of the continental shelf region of the Canadian Arctic.

10:00–10:15

Break

10:15

8UW7. Cramer-Rao bound characterization of equivalent horizontal aperture achieved from environmental asymmetry. John Glattetve (Norwegian Defence Res. Establishment, P. O. Box 115, N-3191

Horten, Norway), John S. Perkins, and W. A. Kuperman (Naval Res. Lab., Washington, DC 20375-5000)

The Cramer-Rao bound is a well-accepted lower bound on the mean-square error of estimated values. Applied to estimation of source

position parameters as measured by a hydrophone array, it will give the least achievable mean-square errors of source position. As is also well known, estimation accuracy of a source's angular position relative to an array is closely connected to the array's aperture transverse to the line of propagation between source and array. By application of the classical optics limit of the Cramer-Rao bound [A. B. Baggeroer *et al.*, J. Acoust. Soc. Am. **83**, 571-587 (1988)] it becomes possible to transform a computed lower bound on angular uncertainty into an equivalent array aperture. It has recently been demonstrated [J. S. Perkins and W. A. Kuperman, J. Acoust. Soc. Am. **87**, 1553-1556 (1990)] that a vertical array deployed in an azimuthally asymmetric environment will achieve angular resolution. Hence, the quantification of this relation between the environment and equivalent horizontal aperture via Cramer-Rao bounds is desired. Numerical modeling results indicating a vertical array's equivalent horizontal aperture will be presented.

10:30

8UW8. Application of sound to underwater acoustics. R. Pitre, Timothy L. Krout, John S. Perkins, B. Edward McDonald, W. A. Kuperman, Michael D. Collins, Nicholas C. Makris (Naval Res. Lab., Washington, DC 20375), and G. L. Gibian (Planning Systems Inc., McLean, VA 22102)

Experienced sonar operators can recognize various types of sounds. With the recent development of time-domain underwater acoustics models and computer workstations capable of generating high-quality sounds from digital data, it is now possible to train the human ear to associate a wide variety of sounds with the underlying physics. With sufficient training, it might be possible to perform qualitative inversion problems such as estimating the range of a sound source, determining properties of the ocean sediment from a bottom-reflected signal, and recognizing the source of a scattered signal. Digitally generated sound can also be used in connection with time-domain signal-processing algorithms such as the delay-and-sum beamformer and the simulated annealing optimal beamformer [W. A. Kuperman *et al.*, J. Acoust. Soc. Am. **88**, 1802-1810 (1990)]. The synthesized sounds presented will include propagation to various ranges in different ocean environments, reflection from different types of ocean bottoms, scattering from bubble clouds near the ocean surface, and the application of beamforming to extract signals from interference (including a single speaker in a noisy crowd).

10:45

8UW9. Multiple source behavior of passive synthetic aperture algorithms. Geoffrey S. Edelson (Dept. of Elec. Eng., Univ. of Rhode Island, Kingston, RI 02881) and Edmund J. Sullivan (Code 103, Naval Underwater Syst. Ctr., Newport, RI 02841)

Generally speaking, passive synthetic aperture algorithms involve the calculation of a single phase correction factor from snapshot to snapshot in order to compensate for the spatial movement of the array over time. Within a given frequency bin and for sources in the far field, this approach is shown to be sufficient for the case of any single source and for the case of two completely deterministic (fully coherent) sources. However, for any other scenario, such as one deterministic and one stochastic source, this technique is shown not to result in the desired phase correction factor. This is a direct consequence of the algorithms used in the various passive synthetic aperture techniques. After reviewing these techniques, an approach to the resolution of this problem, which is based on the multivariate maximum likelihood method, is presented.

11:00

8UW10. Channel-adaptive matched filter processing of large time-bandwidth product signals: Preliminary results. Jean-Pierre Hermand (SACLANT Undersea Res. Ctr., APO, New York, NY

09019) and William I. Roderick (Naval Underwater Syst. Ctr., New London, CT 06320)

Techniques that incorporate propagation channel modeling into conventional matched filter processing can improve detection and localization performance of low-frequency active sonar systems. In January 1989, an acoustic transmission experiment under known oceanographic conditions was conducted by SACLANTCEN in a deep water (2800-m) area west of Sardinia. Large time-bandwidth product signals were transmitted by a broadband projector and were received at a 30-km range in the broadside beam of a horizontal line array of 64 hydrophones. Source and receiver were towed near the channel axis at depths of 100 and 150 m, respectively. The transmitted signals consisted of linear FM pulses with time-bandwidth products that ranged from 400 to 4000. Environmental parameters, calculated from measured and archival data, were introduced in the GENERIC sonar model to compute the eigenrays at discrete frequencies closely spaced over the transmission bandwidth. The (band-limited) impulse response of the channel was obtained by complex summation of the eigenrays in both time and frequency domains. For each bandwidth, the transmitted signal, convolved with the modeled channel response, was correlated against its replica. The predicted matched filter outputs were compared with those obtained from the observations. The results will be discussed in the context of channel-adaptive matched filtering.

11:15

8UW11. Use of bispectral analysis in flow noise measurements. Mary F. Leibolt and Theodore M. Farabee (Code 1942, Ship Acoust. Dept., David Taylor Res. Ctr., Bethesda, MD 20084)

Characterization of the wall pressure field in a turbulent boundary layer is necessary for accurate models to be developed. Measurements of the wall pressure field of a turbulent boundary layer were made in a quiet wind tunnel with flush-mounted pressure sensors. Bispectral analysis was performed on the resulting digitized data to determine whether frequency phase coupling was present. Bispectral analysis, which is an extension of the commonly used Fourier analysis techniques known as auto- and cross-spectral analysis, identifies three wave phase couplings between frequencies by calculating the expected value of the sum of the phases of three frequencies of interest:

$$B(f_1, f_2) = E\{|X(f_1)| |X(f_2)| |X(f_1 + f_2)|\} \\ \times \exp[i\theta(f_1) + \theta(f_2) + \theta(f_1 + f_2)],$$

where $|X(f)|$ and $\theta(f)$ are the respective magnitude and phase of the Fourier transform of $x(t)$. Phase coupling can be indicative of nonlinear relationships in a system. [Work supported by DTRC IR/IED Program.]

11:30

8UW12. Evaluation of interference effect in the estimation of underwater target abundance by the echo-integration method. Zhigang Sun and Gérard Gimenez (Lab. "Traitement du Signal et Ultrasons," U.R.A. C.N.R.S. 1216, INSA, Bât. 502, 69621 Villeurbanne Cedex, France)

The estimation of fish abundance by the echo-integration method is based on the assumption that the total integrated echo intensity returned from randomly distributed targets is proportional to their quantities. In the case where the multiple scattering between targets is negligible, the precision of this estimation can be affected by the interference between individual echos from every single target. In this work, an evaluation function to determine the eventual divergence of the estimated result from the linearity is proposed. As a function of different parameters, such as the total number and the dimension of targets, the transmitted signal frequency, distance from transducer to the network plan and the transducer beamwidth, the importance of interference effect is analyzed for a network of spheres with various spatial distributions. [Work supported by IFREMER.]

Session 9BV

Bioresponse to Vibration: Tactile Pattern Perception: Psychophysics and Neurophysiology

Roger W. Cholewiak, Chair

Psychology Department, Princeton University, Princeton, New Jersey 08544-1010

Chair's Introduction—1:00

Invited Papers

1:05

9BV1. Expansions and modifications of the four-channel model for touch. S. J. Bolanowski (Inst. for Sensory Res., Syracuse Univ., Syracuse, NY 13244)

The model proposing that there are four distinct channels mediating the mechanical aspects of touch [Bolanowski *et al.*, *J. Acoust. Soc. Am.* **84**, 1680–1694 (1988)] was based on psychophysical and physiological experiments conducted on the glabrous skin of humans and required assumptions regarding the neural code used by each channel. Continuing psychophysical experiments have expanded the model to include hairy skin which, in the least, operates using three separate channels: a low-frequency (0.4–3 Hz) one that is temperature insensitive, and middle- (3–50 Hz) and high- (50–500 Hz) frequency ones that are both sensitive to temperature. The sensitivity of all of these channels is greater for larger stimulus areas. Furthermore, physiological experiments directed at testing some of the assumptions used in the model to establish the psychophysical–physiological link show that it must be modified. In particular, activity arising from a single Pacinian corpuscle cannot account for the increases in sensitivity that occur for increases in stimulus duration required by the phenomenon of temporal summation known to exist in the P channel. Therefore, it is likely that the neural code for threshold in the P channel requires the activation of more than one Pacinian corpuscle. This finding may have implications regarding the neural code used by other channels possessing temporal summation (e.g., NP II).

1:30

9BV2. Vibrotactile adaptation. Mark Hollins, Alan K. Goble, and Kimberly A. Delemos (Dept. of Psychol., Univ. of North Carolina, Chapel Hill, NC 27599)

Perception of a vibratory test stimulus is substantially influenced by the level of vibrotactile stimulation that precedes it. In a series of experiments reported here, effects of such adaptation on detection and discrimination thresholds were examined. All stimuli were sinusoidal vibrations delivered perpendicular to the skin through a flat, circular contactor; thresholds were measured during "probe" test periods separated by adapting periods. Absolute thresholds showed a negatively accelerated rise following onset of the adapting stimulus, on both the hand and the face. In contrast to the fact that receptor channels serving the hand adapt independently of one another [R. T. Verrillo and G. A. Gescheider, *Sens. Proc.* **1**, 292–300 (1977)], some evidence was found for channel interaction on the face, a result implying central nervous system (CNS) involvement. Amplitude difference thresholds, measured on the hand with two-interval forced-choice tracking, decreased following exposure to adapting stimuli comparable in amplitude to the test stimuli; this finding is consistent with data [G. A. Gescheider and G. H. Wright, *J. Exp. Psychol.* **77**, 308–313 (1968)] showing that the psychophysical function is steeper after adaptation. A recently proposed model of somatosensory cortical dynamics [B. L. Whitsel *et al.*, in *Information Processing in the Somatosensory System*, edited by O. Franzen and J. Westman (MacMillan, London, 1990)] offers a plausible physiological context for understanding these results. [Work supported by USPHS Grant No. DE-07509.]

1:55–2:00

Break

2:00

9BV3. Cortical representations of learned vibrotactile stimuli. M. M. Merzenich (Coleman Lab., and Depts. of Otolaryngol. and Physiol., Univ. of California, San Francisco, CA 94143-0732)

Recent electrophysiological studies have shown that cortical somatosensory representations are shaped by tactual experiences. These adaptive neuronal processes underlying cortical contributions to tactual learning and nondeclarative memory have been studied by: (a) distorting the tactual experiences of monkeys and rats over a limited time epoch; (b) determining the consequences of special skin and peripheral nerve manipulations that test hypotheses about the neural network origins and cortical plasticity; (c) evaluating the consequences, for cortical representation and cortical network organization, of the perturbation of the network by microstimulation; and (d) training monkeys to make distinctions about tactile stimuli while tracking training-induced changes in their cortical representations. These studies bear significance for understanding the neural origins of tactile perception, and provide a basis for understanding how idiosyncratic tactual perceptual abilities emerge from tactile experience. [Work supported by NIH Grant NS-10414, the Coleman Fund, and Hearing Research, Inc.]

2:25

9BV4. Altering tactile spatial sensitivity. James C. Craig (Dept. of Psychol., Indiana Univ., Bloomington, IN 47405)

Recent animal studies have demonstrated that extended tactile experience can alter cortical organization [Jenkins *et al.*, *J. Neurophysiol.* **63**, 82-104 (1990)]. The present study examined the effect of extended tactile stimulation on spatial sensitivity in human subjects. Four subjects received repetitive, tactile stimulation presented on the volar surface of the forearm. The tactile stimuli, taps, were delivered periodically through small vibrators worn by the subjects for up to 18 h per day. After a period of 5 to 8 weeks of wearing the vibrators, three of the four subjects reported anomalous sensations when attempting to localize tactile stimuli. Subjects had difficulty in localizing stimuli and reported sensations of pressure and diffuseness. Single stimuli applied to the forearm would sometimes evoke double and triple sensations separated by as much as 20 cm. After removing the vibrators, subjects continued to report anomalous sensations for up to 15 weeks. These results suggest that the neural organization of human subjects may be altered by extended tactile experience.

2:50-3:00

Break

3:00

9BV5. Neural mechanisms of tactual roughness perception. Kenneth O. Johnson, Charles E. Connor, and Steven S. Hsiao (Dept. Neurosci., Johns Hopkins School of Medicine, 725 N. Wolfe St., Baltimore, MD 21205)

The neural mechanisms underlying tactual roughness were investigated in a combined psychophysical and neurophysiological study. Stimuli consisted of surfaces embossed with dot arrays of varying dot diameter and spacing. Human subjects scanned the surfaces tactually and responded with numerical magnitudes proportional to their sense of roughness magnitude. The same surfaces were scanned across the receptive fields of cutaneous mechanoreceptive afferents in monkeys while recording the evoked action potentials. Hypothetical neural codes for roughness magnitude were computed from the neural response patterns and tested for their ability to account for the psychophysical data. Four types of neural coding mechanisms were considered: (1) mean firing rate; (2) general variation in firing rate; (3) short-term temporal variation in firing rate; and (4) local spatial variation in firing rate. Mean firing rate failed to explain the psychophysical results: Surfaces that evoked the same firing rate evoked very different roughness judgments. In contrast, neural codes based on spatial firing rate variation, especially in slowly adapting afferents, account for the psychophysical results [Connors *et al.*, *J. Neurosci.* **10**, 3823-3826 (1990)]. [Work supported by NIH.]

3:25

9BV6. Representation of spatial patterns on the Optacon in the peripheral and central nervous system. E. P. Gardner, C. I. Palmer, H. A. Hamalainen, and S. Warren (Dept. of Physiol. and Biophys., New York Univ. Medical Ctr., 550 First Ave., New York, NY 10016)

The pattern of neuronal activity evoked by the Optacon stimulator has been defined for cutaneous mechanoreceptors and for cortical neurons using bar patterns scanned across the tactile array. Pulses of 4

ms were delivered at 25–100 Hz in a row-by-row sequence stimulating motion of a horizontal bar along the finger. These stimuli activated only phasic mechanoreceptors (RA and PC afferents) but not slowly adapting afferents. RA afferents view only a small fraction of spatial patterns, as they respond to only 2–4 adjacent rows, firing one spike/pulse. Responses are all or none, and constant throughout the period of stimulation. Resolution of stripe spacing appears limited by the receptive field diameter. Total spike output is constant at all frequencies tested, and average firing rates mimic the stimulation frequency. Central neural networks transform the peripheral input such that S-I cortical neurons show a linear rise in total spike output and constant average firing rates as the interpulse interval is lengthened from 10–40 ms. Firing is higher than the stimulation rate at 25 Hz, and reduced to half the input frequency at 100 Hz, suggesting amplification at low Hertz and inhibition at high Hertz. Stimulation rates are represented by the pattern of firing within the spike train, but the modulation amplitude is frequency dependent. It is suggested that the low-amplitude continuous firing observed at 100 Hz correlates with sensations of smooth motion across the finger, while the large fluctuations in neural excitability at 25 Hz yield distinct, punctate sensations. [Work supported by NS11862.]

3:50–4:00

Break

Contributed Papers

4:00

9BV7. Identifying the direction of simulated movement on the skin: The effects of an irrelevant stimulus. Paul M. Evans (Dept. of Psychol., 900 State St., Willamette Univ., Salem, OR 97301) and James C. Craig (Indiana Univ., Bloomington, IN 47405)

Movement was simulated on the index and middle fingerpads by activating, in rapid succession, adjacent columns (or rows) of the tactile display of the Optacon. The stimuli either moved from left to right (or vice versa), or from the top of the display to the bottom (or vice versa). Subjects were trained to respond "1" for two of the stimuli and "2" for the remaining stimuli. The subject's task was to focus attention on the index fingerpad (the target location), to identify the stimulus that was presented to that site, and to ignore the stimulation on the middle fingerpad (the nontarget location.) There were three trial types: (1) the stimuli were physically identical (moved in the same direction); (2) the stimuli were physically different but assigned the same response; and (3) the stimuli were different and assigned different responses. The results showed that when the nontarget and target had the same response (regardless of whether they were physically identical or not), accuracy was higher and reaction times were faster than when the nontarget and target had different responses. The results suggest that subjects are unable to restrict processing to a single site on the hand. Moreover, a tactile stimulus at a nontarget location appears to be processed to the level of response activation. [Work supported by NIH.]

4:15

9BV8. Judgments of tactile texture gradient magnitude. Gunnar Jansson (Dept. of Psychol., Uppsala Univ., Box 1854, S-751 48 Uppsala, Sweden) and Barry Hughes (Univ. of Illinois at Chicago, Chicago, IL 60680)

A main problem in reading tactile pictures is the perception of depth. In visual pictures, texture gradients are very effective in providing 3-D information. The aim of this study was to investigate one aspect of this potential source of information in tactile pictures, namely, judgment of the magnitude of tactually presented texture gradients. The stimuli consisted of polar projections of regular plane patterns of points

and lines at a slant from the frontal plane. Such projections were copied onto swell paper which, after heating, provided the texture gradient in imbossed form. The patterns were read with the tip of the index finger without any restrictions concerning kind of exploratory movements. The result was that the tactile texture gradients were judged well in accordance with the physical magnitude of the gradient. Other experiments, investigating hypotheses derived from analyses of recordings of the exploratory movements, with stimuli containing either the whole gradient, its central part, or its extremes demonstrated very similar results for the whole gradient and its extremes. The theoretical significance of these results is discussed.

4:30

9BV9. Vibrotactile thresholds for detection of sinusoidal vibration as a function of stimulus duration measured in the presence of vibratory background noise. G. A. Gescheider (Hamilton College, Psychol. Dept., Clinton, NY 13323 and Syracuse Univ., Inst. for Sensory Res., Merrill Lane, Syracuse, NY 13210-5290), Kathleen Hoffman, Michael Travis (Hamilton College, Clinton, NY 13312), Stanley J. Bolanowski, Jr., and Ronald T. Verrillo (Syracuse Univ., Syracuse, NY 13210-5290)

Thresholds for the detection of sinusoidal vibration were measured for stimuli ranging in duration from 15 to 1000 ms. The test stimuli were 250-Hz bursts of vibration with 10-ms rise-fall times applied through either a 3.0 or 0.01 cm² contactor to the thenar eminence of the right hand. Thresholds were measured in the presence of and in the absence of narrow-band noise with frequencies centered around that of the test stimuli. Temporal summation, as indicated by a decrease in thresholds as stimulus duration increases, was observed at all intensities of the noise masker. This was true whether the stimulus was delivered through a large contactor designed to stimulate the Pacinian channel or through the small contactor designed to stimulate non-Pacinian systems. On the other hand, when stimuli were detected in the presence of sinusoidal maskers, the amount of temporal summation depended on the intensity of the masker in a way predictable from the hypothesis that temporal summation can occur in the Pacinian, but not in one of the non-Pacinian channels.

Session 9MU

Musical Acoustics: Glass Musical Instruments

Thomas D. Rossing, Chair

*Department of Physics, Northern Illinois University, DeKalb, Illinois 60115**Invited Papers*

1:00

9MU1. Acoustics of the glass harmonica. Thomas D. Rossing (Dept. of Physics, Northern Illinois Univ., DeKalb, IL 60115)

Modern glass harmonicas have developed along two different lines: One type consists of a collection of wineglasses, which the player sets into vibration by moving a finger around the rim. The other type, called the armonica by its inventor Benjamin Franklin, consists of a series of glass bowls mounted on a horizontal shaft so that they rotate together. The vibrational modes of a glass harmonica are quite similar to those of a bell, the fundamental (2,0) mode determining the pitch. The tangential component of motion in this mode (which is 1/2 the normal component) is apparently excited by stick-slip coupling to the moving finger of the player. Simple theory predicts that the frequency will increase as $1/r^2$, where r is the radius of the glass, and t is its thickness [T. D. Rossing, *Phys. Teacher* **28**, 582–585].

1:30

9MU2. The glass harp: "The sound of heaven." Jamey Turner (3308 Holloman Rd., Falls Church, VA 22042)

This presentation will consist of three parts: (1) A musical performance on the glass harp will demonstrate the incredible beauty of this instrument, which the author has played on all the national television networks and with many symphony orchestras. Music written for several kinds of glass instruments played by rubbing will be performed, including some by Mozart and Beethoven. Glasses will be distributed for audience participation on one selection. (2) A short historical discussion will be given of the rich legacy of glass instruments and music spanning three centuries, with roots that go back over 2000 years to glass instruments tapped rather than rubbed. (The author was privileged to examine the original glass harmonica invented by Ben Franklin.) (3) The making of the glass harp will be discussed, with emphasis on selecting the glasses, the water, the table, and other elements that are important to the quality of the sound.

2:00

9MU3. The glass harmonica: A return from obscurity. Gerhard Finkenbeiner, Vera Meyer, and Alisa Nakashian (33 Rumford Ave., Waltham, MA 02154)

The authors introduce the glass harmonica by placing it in its historical context. It is explained how, after more than a century of obscurity, the glass harmonica has been recreated using updated methods and materials. Specific features of the instrument are examined, including its adaptability to a wide range of compositional works.

2:30

9MU4. The Benjamin Franklin glass harmonica. Kathleen J. Allen (Dept. of Humanities, Music and Theater Arts Sec., MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

The instrument described is from a design by Benjamin Franklin, of modern manufacture by Gerhard Finkenbeiner of Waltham, Mass., played by Vera Meyer of Cambridge, Mass., and encompasses three octaves. Two notes from each octave were investigated at varying amplitudes of excitation and different speeds of rotation. Data will be presented to show that harmonic richness is a function of octave, and is relatively unaffected by amplitude of excitation; also that the rotation of the disks splits the degeneracy of the normal modes. An analysis will be presented of the splitting by decomposition into traveling waves around the circumference of the disk, which relates rotation speed and the shear velocity of sound in the glass medium.

9MU5. The glass flute as an evolutionary step between wood and metal. Robert E. Sheldon (Curator of Musical Instruments, Library of Congress, Washington, DC), Sara Stern (2123 N. Oakland St., Arlington, VA), and Edith L. R. Corliss (Forest Hills Lab., 2955 Albemarle St. NW, Washington, DC 20008)

In the course of development of flute tone, the material of which the flute was made became an important consideration. The upper harmonic structure is affected both by the shape of the instrument and by the material of which it is made. Modern flutes are made of dense and rigid metals. Along the way, a few 19th-century instrument makers made flutes of glass. The physical properties of glass flutes will be discussed and demonstrated.

FRIDAY AFTERNOON, 3 MAY 1991

CARROLL, 1:00 TO 3:30 P.M.

Session 9PA

Physical Acoustics: General Topics

James B. Mehl, Chair

Department of Physics and Astronomy, University of Delaware, Newark, Delaware 19716

Contributed Papers

1:00

9PA1. Large amplitude wall pressure events beneath turbulent boundary layers. Carolyn Karangelen (Dept. of Mech. Eng., Catholic Univ. of America, Washington, DC 22064)

Examination of the fluctuating wall pressures can yield much information on the turbulent structures within a flow. The wall pressure signal may be considered a footprint of these structures. The pressure fluctuations measured at the wall are produced from velocity fluctuations within the flow that are propagated through the flow medium. The sporadic large amplitude wall pressure events are thought to be footprints of turbulent bursting events in the near wall layer. Large amplitude events were captured using conditional sampling techniques and analyzed for their statistical, temporal, and spectral characteristics. The results of this analysis will be presented. [Funded by ONR and IBM.]

1:15

9PA2. Interior pressure field in an accelerated spherical container. Christopher L. Morfey^{a)} (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton SO9 5NH, U.K.)

The pressure field inside a rigid hollow container, filled with uniform fluid and subjected to an impulsive translational acceleration $a(t)$, is examined analytically by time-domain methods. In the incompressible limit, a uniform pressure gradient $-\rho a(t)$ exists throughout the interior. However, compressibility introduces acoustic wave phenomena, which are investigated in detail in this paper for a spherical container. The exterior version of the problem is discussed by Pierce, and a solution given, in his book [A. D. Pierce, *Acoustics* (1982)]. The present interior problem has received some attention in the biomechanics literature, as a step toward modeling the dynamics of closed-head injury: In this context, it describes events on a short time scale, of order 20 μ s

rather than 20 ms (which is the typical impact duration in blunt head trauma). ^{a)}On leave 1990/91 at Pennsylvania State University, Ctr. for Acoust. and Vibration, and Univ. of Texas at Austin, Appl. Res. Lab., Austin, TX 78713-8029.

1:30

9PA3. Ultrasonic focus degradation produced by abdominal wall. Robert C. Waag (Dept. of Elec. Eng., Univ. of Rochester, Rochester, NY 14627)

Focus degradation produced by human abdominal wall has been studied experimentally using measurements of ultrasonic pulses from a special curved transducer that emits a hemispheric wave and simulates a point source. The pulse waveforms were measured in a two-dimensional aperture after propagation through a water path and after propagation through five different specimens of human abdominal wall. The time history of the virtual point source was reconstructed from the measurements by removing the time delay produced by geometric path differences, finding the complex amplitudes of the temporal harmonics across the aperture, calculating the Fraunhofer diffraction pattern of each harmonic, and summing the patterns. Comparison of the time history in the plane of the virtual source reconstructed from signals that propagated through the abdominal wall specimens and the time history obtained from signals without the abdominal wall specimens present shows that the main features of the reconstructed source waveform are relatively unaffected by the presence of the human abdominal wall but that significant differences in signals greater than 40 dB down from the peak result from propagation through the specimens employed in this study.

9PA4. Excitation of resonators by blood flow in arteries. T. Douglas Mast (Graduate Prog. in Acoust., Penn State Univ., Appl. Sci. Bldg., University Park, PA 16802)

Resonators are known to exist in human arteries. These include aneurysms and weakened sections of the arterial wall. The present paper gives a quantitative theory of how such resonators are excited under physiological conditions. Considered mechanisms of excitation include turbulence and flutter. The pulsatile nature of blood flow is taken into account, as are the flexible walls of arteries. The time-varying frequency, bandwidth, and amplitude of such resonances are related to parameters that characterize the cardiovascular system. [Work supported by the William E. Leonhard endowment to Penn State Univ. The author acknowledges the advice of A. D. Pierce.]

2:00

9PA5. Spherical cavity resonator: Singular boundary-shape perturbation? James B. Mehl (Physics Dept., Univ. of Delaware, Newark, DE 19716)

A nearly spherical cavity resonator has been investigated using boundary-shape perturbation theory. The cavity parts consist of two perfect hemispheres of radii a and $b = a(1 - \epsilon)$, aligned along their common axes and connected by a plane surface at $\theta = \pi/2$. The lowest-frequency nonradial modes with acoustic pressure proportional to $\Phi_z = j_1(kr)\cos\theta$ and $\Phi_x = j_1(kr)\sin\theta\cos\phi$ have been investigated. These modes have fairly uniform velocity fields oscillating along the z and x axes, respectively. Boundary-shape perturbation theory has been applied to develop an expression for the eigenfrequency perturbation in powers of the small parameter ϵ . The first-order perturbations of the eigenfrequencies vanish for both modes. The second-order perturbation series for the Φ_x mode converges; the results agree with numerical calculations based on a boundary-integral-equation (BIE) technique. The second-order perturbation series for the Φ_z case diverges. The BIE numerical results suggest the presence of a nonsingular term proportional to $\epsilon^2 \log \epsilon^2$.

2:15

9PA6. A simple circuit model for the thermodynamics of thermoacoustic devices. Peter H. Ceperley (Depts. of Electrical and Computer Eng. and Physics, George Mason Univ., Fairfax, VA 22030)

A simple circuit model of the thermodynamics of thermoacoustic devices will be presented. This model allows the simple calculation of thermoacoustic gain, efficiency, and thermal current. More importantly, it allows a simpler, more intuitive approach to optimizing the parameters of a thermoacoustic device. Results calculated using this model will be compared with the exact results calculated for a parallel plate (or slit) geometry for a range of slit widths and phasings. [Work supported by ONR.]

2:30

9PA7. Thermoacoustic properties of porous media. Alon Koren and Peter H. Ceperley (Depts. of Electrical and Computer Eng. and Physics, George Mason Univ., Fairfax, VA 22030)

The thermoacoustic time constant and flow resistance were measured for steel wool, sand, and a parallel plate geometry over a range of frequencies in one atmosphere of air. The results are compared with that of the theoretical parallel plate geometry. Suitability of various packings for thermoacoustic applications will be discussed. [Work supported by ONR.]

9PA8. Complex eigenfrequency analysis of thermoacoustic heat engines. W. Pat Arnott, Richard Raspet, and Henry E. Bass (Natl. Ctr. for Physical Acoust. and the Dept. of Physics and Astron., P.O. Box 847, University, MS 38677)

Gas in a straight tube, open at the cool end and closed at the hot end, can be made unstable with respect to acoustic oscillation by placing a sufficiently large temperature gradient along the tube. The possibility of spontaneous oscillation exists when the externally applied temperature gradient is larger than the temperature gradient associated with a standing acoustic wave in the tube. Energy for oscillation in this non-equilibrium system is supplied by the heat input necessary to maintain the temperature gradient. Rott and his students [Rott, *Adv. Appl. Mech.* **20**, 135-175 (1980)] have investigated this instability for a variety of conditions. The instability analysis is given in this paper for a thermoacoustic prime mover known as a Höffler tube. The complex eigenfrequency (CEF) of a recently constructed Höffler tube is computed as a function of temperature. The frequency of oscillation and the quality factor, or Q , are determined from the real and imaginary parts of the CEF. Spontaneous oscillation can occur in the limit $1/Q \rightarrow 0$. Application of the CEF to thermoacoustic refrigerators is discussed. [Work supported by ONR.]

3:00

9PA9. Observation of a second soundlike mode in superfluid-filled aerogel. M. J. McKenna, Tania Slawacki, and J. D. Maynard (Dept. of Physics, Penn State Univ., University Park, PA 16802)

Superfluid ^4He is interesting acoustically because it can support more than one mode of sound propagation, and these modes may be used in combination to determine the critical properties of the superfluid. Recently, there has been considerable interest in the critical behavior of superfluid ^4He in the presence of a random potential field provided by highly porous media, specifically, silica aerogels whose porosities exceed 90%. Unlike other porous media, where the normal fluid is clamped to the rigid matrix, these aerogels are highly compliant, and while the normal fluid remains locked to the aerogel matrix, both the matrix and the normal fluid can move in response to mechanical and thermal gradients. Therefore, one would not observe ordinary fourth sound but rather a sound mode intermediate between first and fourth sound. In addition, the superfluid can move in a direction opposite to the normal fluid/aerogel matrix, resulting in a second soundlike mode. The first experimental observation of this second soundlike mode in superfluid-filled aerogel is reported, and it is shown that it remains a high-quality mode near the critical temperature. Also presented are measurements of the sound mode intermediate between first and fourth sound, and a theoretical model that gives good agreement with the observed new sound modes. [Work supported by NSF Grant DMR 9000549 and the Office of Naval Research.]

3:15

9PA10. Optimization of spatial resolution and penetration depth using a waveguide in pulsed Doppler ultrasound systems for measurement of blood flow velocity. James J. Finneran and Mardi C. Hastings (Dept. of Mech. Eng., Ohio State Univ., 206 W. 18th Ave., Columbus, OH 43210)

Doppler ultrasound is a noninvasive method used to examine anatomy and various physiological functions. A limitation in its use to measure blood-flow velocity is the size and location of the volume within the blood vessel from which the signal is received. The volume is primarily a function of the size of the ultrasonic transducer and signal frequency. In this study, the feasibility of using a waveguide to mechanically adjust the natural focus of the acoustic field is considered. The adjustable focus allows control of spatial resolution and penetration depth of the ultrasonic signal. By controlling these parameters rather than totally relying

on digital signal processing of received signals, the Doppler system may be used to examine blood vessels of variable size and shape. Measurements of beam profiles in a wave-guided system are compared with

theoretical predictions. Preliminary design of a waveguide for a probe to measure the blood flow velocity at different points through the aorta is presented. [Work supported by NSF.]

FRIDAY AFTERNOON, 3 MAY 1991

INTERNATIONAL C, 1:00 TO 2:30 P.M.

Session 9PP

Psychological and Physiological Acoustics: Auditory Perception in Hearing-Impaired Listeners

Janet Koehnke, Chair

Department of Communication Sciences, University of Connecticut, Box U-85, Storrs, Connecticut 06269

Contributed Papers

1:00

9PP1. Temporal resolution of frequency-modulated signals by hearing-impaired listeners. John P. Madden (Dept. of Speech and Hearing, Cleveland State Univ., Cleveland, OH 44115) and Lawrence L. Feth (Ohio State Univ., Columbus, OH 43210)

Hearing-impaired and normally hearing subjects were asked to discriminate between two sinusoidal signals. One signal, the glide, moved from its initial frequency over a linear path to its final frequency. The other, the step signal, was the same except that its trajectory followed a series of discrete steps in frequency. As the number of steps increased, the duration of the individual steps decreased, and the signal more closely resembled the glide. The center frequencies of the signals were 0.5, 1.0, 2.0, and 4.0 kHz. The signals were presented to the two groups at equal SPLs. The impaired subjects exhibited significantly poorer discrimination than the normally hearing subjects, indicating a reduced ability to temporally resolve the step modulation. A frequency effect was evident in both groups, with much poorer resolution at 4.0 kHz. A level effect was noted in the normal subjects, who exhibited poorer resolution at higher SPLs. The results from the normally hearing subjects were very similar to temporal resolution values obtained in previous studies using amplitude-modulated (gapped) sinusoids. [Work supported by a grant from AFOSR.]

1:15

9PP2. Frequency selectivity and pitch discrimination in young and elderly subjects with cochlear hearing loss. Robert Peters (Speech and Hearing Sci., Dept. of Med. Allied Health Professions, Univ. of North Carolina, Chapel Hill, NC 27514) and Brian C. J. Moore (Dept. of Experimental Psychol., Univ. of Cambridge, Cambridge CB2 3EB, England)

Pure and complex tone pitch discrimination data were obtained for two groups of hearing-impaired subjects, young and elderly, and for normally hearing subjects. Auditory filter shapes were also estimated for center frequencies of 100, 200, 400, and 800 Hz using a modified notched-noise method [B. J. Glasberg and B. C. J. Moore, *Hear. Res.* 47, 103-138 (1990)]. Frequency DLs for pulsed tones were measured for frequencies from 50-4000 Hz. DLs for the fundamental frequency (F_0) of complex tones were measured for F_0 s of 50, 100, 200, and 400

Hz, for complexes containing harmonics 1-12, 6-12, 4-12, and 1-5. The components were added in either cosine phase or in alternating sine-cosine phase. Auditory filters for the young and elderly impaired subjects were similar; both groups had broader filters than the normal subjects. Complex tone DLs were larger for the impaired subjects. Complex tone DLs were especially large for the tones with harmonics 1-5 and 1-12 at F_0 s of 50 and 100 Hz. These DLs were reduced (i.e., performances improved) when the lower harmonics were removed. Complex tone DLs were affected by the relative phases of the components for some but not all of the hearing-impaired subjects. The implications of the results for pitch theories will be discussed. [Research supported by the Andrus Foundation.]

1:30

9PP3. Modulation detection as an index of residual auditory function. C. Formby (Dept. of Otolaryngol., Johns Hopkins Univ. School of Medicine, Carnegie 442, Baltimore, MD 21205), L. Morgan, J. Burton (Univ. of Florida, Gainesville, FL 32611), and T. G. Forrest (Univ. of Mississippi, University, MS 38677)

Indirect tests of residual auditory function often assume that the temporal envelope is the primary cue in the acoustic signal for profoundly hearing-impaired persons who are without auditory function. In contrast, for hearing-impaired persons who retain residual auditory function, both temporal envelope and spectral cues are available. This hypothesis was evaluated directly by studying simultaneously temporal envelope and spectral resolution in seven profoundly hearing-impaired subjects (ten ears). Amplitude modulation (AM) detection thresholds were measured with a 250-Hz carrier, modulated at rates of 40 and 150 Hz, presented by headphones and by hand vibration. At 40 Hz, where the AM sidebands fell within the same critical band as the carrier and could not be resolved, all subjects yielded similar headphone and vibrator results. At 150 Hz, headphone thresholds for six ears were better than the vibrotactile threshold, while four ears yielded little or no difference in performance between the two transducers. The latter results reflect differences in resolution between ears with and without functional critical band mechanisms and, hence, inherent differences between auditory and vibrotactile processing. [Research supported by NIH.]

9PP4. Inaudible stimulus may cause threshold shift. I. M. Young and L. D. Lowry (Dept. of Otolaryngol., Jefferson Med. College of Thomas Jefferson Univ., Philadelphia, PA 19107)

Interrupted pure-tone threshold measurements were made monaurally by Békésy audiometry with and without presentation of inaudible identical continuous stimulus and white noise on subjects with bilateral normal hearing and unilateral sensorineural hearing loss with and without abnormal auditory threshold adaptation. Thresholds were compared for interrupted stimulus only and both interrupted stimulus and inaudible stimulus presentation. Subjects with normal hearing and sensorineural hearing loss without abnormal adaptation showed no difference with and without presentation of the inaudible continuous stimulus. Subjects with sensorineural hearing loss demonstrating marked abnormal adaptation revealed observable threshold shift for the interrupted stimulus by adding inaudible stimulus. It is assumed that the inaudible stimulus affecting partially damaged nerve fibers produce widespread abnormal adaptation that the threshold for another signal is inaudible until an abnormally high intensity is reached. A stimulus incapable of evoking a response may cause adaptation in such fibers.

2:00

9PP5. Auditory factors in obscure auditory dysfunction. Susan R. Mahanes and Robert Peters (Speech and Hearing Sci., Dept. of Med. Allied Health Professions, Univ. of North Carolina, Chapel Hill, NC 27599-7190)

Obscure auditory dysfunction (OAD), defined as a self-reported difficulty in understanding speech in noise by persons with normal audiograms and no other obvious causes, is commonly seen in clinical settings [G. H. Saunders and M. P. Haggard, *Ear Hear.* 10, 200-208]. In an attempt to characterize the hearing difficulties experienced by this population, measures of frequency selectivity, complex and pure-tone

discrimination and gap detection were obtained, as well as speech, audiometric, and case history data. Results indicated impaired frequency selectivity at low frequencies, especially at 100 Hz and higher than normal thresholds for complex- and pure-tone pitch and gap detection also at the lower frequencies. Speech reception thresholds in noise were not consistently higher than for normal listeners. [Research supported by the American Speech-Language-Hearing Foundation, NC Regional Chapter of the Acoustical Society of America, and the Andrus Foundation.]

2:15

9PP6. Monaural detection with a contralateral cue in normal-hearing and hearing-impaired listeners. J. Koehnke and J. Besing (Univ. of Connecticut, Dept. of Commun. Sci., Box U-85, Storrs, CT 06260, and MIT, Cambridge, MA 02139)

Monaural detection with and without a contralateral cue (MDCC) in normal-hearing and hearing-impaired listeners is investigated. The signals and cues are 1/3-octave noise bands centered at 500 and 4000 Hz, and the masker is a 4500-Hz low-pass noise. The level of the masker is 77 dB SPL for the normal-hearing listeners and 25 dB SL for the hearing-impaired listeners; when present, the cue is at -7 dB relative to the masker. Psychometric functions for three normal-hearing subjects and three subjects with moderate-to-severe, bilateral sensorineural hearing losses have been measured. The normal-hearing listeners show a 2-8 dB cued advantage at 500 Hz and a 1-11 dB cued disadvantage at 4000 Hz. Like the normal-hearing listeners, the hearing-impaired subjects have poorer performance at 4000 Hz when the cue is present. At 500 Hz, however, one of the hearing-impaired listeners shows a 5 dB cued advantage, comparable to normal, while the other hearing-impaired subjects have poorer detection when the cue is present. Results of these MDCC measurements will be compared with the performance of these hearing-impaired listeners on other tests of binaural detection and discrimination. [Work supported by DRF.]

FRIDAY AFTERNOON, 3 MAY 1991

INTERNATIONAL B, 1:00 TO 4:45 P.M.

Session 9SP

Speech Communication: Word and Sentence Effects

H. Timothy Bunnell, Chair

Speech Processing Laboratory, Alfred I. du Pont Institute, Wilmington, Delaware 19899

Contributed Papers

1:00

9SP1. Underwater audiological testing of a diver communicator. Richard W. Ranlet (Raytheon SSD, Box 360, Portsmouth, RI 02871)

An underwater communicator was developed to evaluate the suitability of direct audio voice transmission as a practical means to enable close-range communications between divers using open-circuit SCUBA. A communicator that transmits speech directly into the water was designed and human engineered to yield a practical working prototype suitable for normal open-water dive conditions. Intelligibility testing of the device entailed underwater audiological testing that attempted to match in-air standards and practices. Measurements of hearing thresh-

olds, speech reception thresholds, and speech discrimination/recognition were performed using accepted ASHA practices. Intelligibility of sentences deemed relevant to the diving environment was also tested. In-water intelligibility testing levels were correlated with accepted in-air standards using measured hearing and speech reception thresholds as common reference points between the two media. Binaural intelligibility tests were performed in a pool using Auditec and Campbell PB50 word lists. Based on averaged results from two underwater listeners, intelligibility for the finished communicator measured 82% correct responses (5-m distance between transmitting and receiving divers). Similar tests for sentence intelligibility yielded near 100% correct responses for the same conditions.

9SP2. The intelligibility of words, sentences, and continuous discourse using the articulation index. Rory DePaolis (Graduate Prog. in Acoust., Penn State Univ., University Park, PA 16803)

The relation between the intelligibility of speech stimulus with widely varying message redundancy was measured. Using the original articulation index (AI) methodology [N. R. French and J. C. Steinberg, *J. Acoust. Soc. Am.* **19**, 90–119 (1947)], frequency importance functions and transfer functions (AI versus percent correct) were determined for one speaker speaking 616 PB-50 words, 200 meaningful SPIN sentences, and 44 7th-grade reading level continuous discourse, (CD) passages. Thirty subjects were instructed to write down each word and to estimate the percentage of each sentence and CD passage that they heard correctly. The stimulus was degraded with 4 noise and 11 filtered conditions. The results demonstrate a trend toward less significance of high-frequency cues as message redundancy increases. There is also evidence to recommend the use of speech type specific frequency importance functions when calculating the AI.

1:30

9SP3. A probability model for distributions of speech intelligibility data. Caldwell P. Smith (Consultant, 378 Chicopee Row, Groton, MA 01450)

It was determined that the compound Poisson probability distribution as described by William Feller in *An Introduction to Probability Theory and Its Applications*, 2nd ed. [Wiley, New York, 1975], pp. 270–273] is a valid model for distributions of speech intelligibility scores from diagnostic rhyme tests. This was established from details of scores from 110 multispeaker tests of a variety of speech processing conditions. Probability models were constructed by first converting feature scores to integers representing frequencies of errors in listener responses, and calculating means and variances of those distributions. Variance of a compound Poisson distribution is equal to the mean divided by p , and in this corpus of data the value of p tended to remain relatively fixed at an average value of 0.129, with the consequence that distributions were essentially defined by mean values and dispersions a linear function of means. In these measures, variance averaged 7.75 times the mean, with the average value of this coefficient varying over a limited range with different speech processing conditions: for LPC processors, the average was 8.37; for wideband processors, 7.12; for processors in tandem, 6.75; and for speech in Gaussian noise, 8.06. Partitioned into separate data sets for voiced and unvoiced feature scores, the same trends were observed, but with coefficients approximately 15% larger with voiced data, and approximately 20% smaller with unvoiced data.

1:45

9SP4. Multimode database and its preliminary results. Karen Wallace, Josephine Horna, Yukiko Monzen, and Noriko Umeda (Dept. of Linguistics ISALS, New York Univ., 719 Broadway, New York, NY 10003)

Data from four speaking modes—word-list reading, text reading, conversation, and reading of sentences occurring in the conversation—were collected from American English speakers (five male, five female). Each speaker spoke for approximately 5 h. Smaller amounts of similar data are being collected for Japanese and Spanish. The principal objective is to better understand speech activities and their underlying rules by capturing them in speaking modes that exhibit very different ranges of variation. The reading of sentences spoken in conversation is particularly useful, because it allows us to directly compare acoustic attributes of linguistic forms in formal versus spontaneous speech. Several research projects that contrast conversation and more formal modes (two of which will be presented at this meeting) are being carried out. These include formant frequencies of unstressed vowels [Wallace]; acoustic properties of /ə/s [Horna]; prosodic attributes of stressed and unstressed vowels; perception and acoustic correlates of degrees of empha-

sis. Ideas for future research, including a cross-language study of contractions used in conversation, will be proposed.

2:00

9SP5. Time course of hemispheric differences in spoken word recognition. Edward T. Auer and Paul A. Luce (Lang. Percept. Lab., Dept. of Psychol., State Univ. of New York, Buffalo, NY 14260)

Previous research has demonstrated differences between hemispheric processing of auditorily and visually presented lexically ambiguous words. The present study further examined the time course of hemispheric processing of lexically ambiguous spoken words. Lexically ambiguous primes (e.g., BANK) were presented binaurally. Fifty or 500 ms later, a target word was then presented monaurally to the left or right ear for a speeded lexical decision response (i.e., WORD–NONWORD). Target words were either (1) related to the dominant meaning of the ambiguous prime (BANK–MONEY), (2) related to the subordinate meaning of the prime (BANK–RIVER), or (3) unrelated to the prime. No significant differential effects of facilitation were found in any condition. In addition, only the fastest subjects in the 50-ms ISI experiment showed a significant right-ear advantage. Implications of these results will be discussed in terms of modality specificity and temporal versus spatial stimulus array.

2:15

9SP6. Speeded comparisons of spoken words. Paul A. Luce (Lang. Percept. Lab., Dept. of Psychol., State Univ. of New York, Buffalo, NY 14260)

This research evaluated the claim that spoken words are recognized as soon as they diverge from all other words in memory. Pairs of consonant–vowel–consonant (CVC) words were presented for a speeded SAME–DIFFERENT response. Subjects were instructed to respond SAME or DIFFERENT as soon as they could determine if the second word of the pair was the same as or different from the first word. Reaction times were measured from the onset of the second word. The overlap between the two members of the pairs of spoken words was systematically manipulated to determine if reaction times to say DIFFERENT were a simple function of the point at which the two words differed or whether overlapping segments after the divergence point would affect reaction times. The results are consistent with the notion that the temporal window for mapping information onto representations of spoken words in memory spans more than a feature or segment and that spoken words are not necessarily recognized as soon as they diverge from other words in memory.

2:30

9SP7. Some lexical effects in a generalized phoneme monitoring task. Scott E. Lively and David B. Pisoni (Speech Res. Lab., Dept. of Psychol., Indiana Univ., Bloomington, IN 47405)

A long-standing concern in the spoken word recognition literature has been whether use of the lexicon is necessary to complete a phoneme monitoring task. Eimas *et al.* [*J. Mem. Lang.* **29**(2), 160–180 (1990)], for example, found no lexical effects in a phoneme monitoring task until monitoring responses were accompanied by lexical decisions or noun–verb categorizations. Frauenfelder and Segui [*Mem. Cog.* **17**(2), 134–140 (1989)], in contrast, found facilitatory priming effects in a monitoring task when the position of the target phoneme varied randomly from trial to trial. The current study adopts Frauenfelder and Segui's generalized phoneme monitoring paradigm in a speeded phoneme classification task. Subjects participating in a blocked condition demonstrated word frequency and density effects only for blocks of trials in which targets occurred word finally. No lexical effects were observed for responses to word initial targets. Subjects in a mixed condition showed frequency and density effects for word final targets and frequency effects for word initial targets. In terms of Cutler's race model, the data indi-

cate that subjects adopt a postlexical response strategy when targets occur late in the stimulus word or when attention cannot be consistently

focused on a particular target position. [Work supported by PHS RO1 DC011-15.]

2:45–3:00

Break

3:00

9SP8. The role of lexical status in the segmentation of fluent speech. Anne S. Henly and Howard C. Nusbaum (Dept. of Psychol., Univ. of Chicago, 5848 S. University Ave., Chicago, IL 60637)

Theories of word recognition propose that listeners use lexical status to segment one word from another in fluent speech. Thus words must be recognized one at a time, in the order in which they were produced. This leads directly to the following predictions: (1) Words should be easier to identify following a word than following a nonword. (2) The lexical status of a syllable following a word should not affect the identification accuracy of that word. Subjects in the present experiment were asked to identify monosyllabic and trisyllabic target words presented in noise. Target words were presented with preceding word and nonword context syllables, as well as following word and nonword context syllables. Although the results confirm that listeners are able to use lexical status to facilitate segmentation, they also strongly suggest that listeners' use of lexical status is quite unlike the segmentation strategies proposed by most models of word recognition. [Research supported by NIDCD.]

3:15

9SP9. Listening to the sound of sentences. Howard C. Nusbaum, Kevin J. Broihier (Dept. of Psychol., Univ. of Chicago, 5848 S. University Ave., Chicago, IL 60637), and Judith C. Goodman (University of California, San Diego, CA)

Although intonation conveys a great deal of information relevant to understanding sentences, it is unknown how listeners actually use this information. How do listeners integrate information in the intonation of a sentence with information derived from a linguistic analysis of the words in the sentence? Syntactic information from the order of words in a sentence may be processed independently from syntactic information perceived from intonation. On the other hand, different sources of syntactic information may be treated as equivalent and integral. Subjects were instructed to judge whether the intonation of a sentence was declarative or interrogative. Statements and questions were produced in two forms; with a declarative intonation and with an interrogative intonation. In one condition, syntactic structure was constant for all trials and intonation varied. In a second condition, syntactic structure varied across trials, as did intonation. The results indicate that listeners are unable to ignore the syntactic structure of a sentence in judging the intonation of the sentence. Listeners treat different types of syntactic information as integral. These results suggest that perception of intonation is a direct and integral part of sentence understanding. [Research supported by NIDCD.]

3:30

9SP10. Some effects of text coherence on the comprehension of natural and synthetic speech. James V. Ralston, Scott E. Lively, and David B. Pisoni (Speech Res. Lab., Dept. of Psychol., Indiana Univ., Bloomington, IN 47405)

Subjects listened to naturally and synthetically produced (Votrax Type-*n*-Talk) passages of varying levels of difficulty in a sentence-by-sentence listening time task. Listeners controlled the intersentence interval while listening to passages presented in their normal sentence order or in a random sentence order. Subjects listening to Votrax speech had significantly longer intersentence response times in both the normally ordered and the randomly ordered sentence conditions. Furthermore, in a recognition test given after each passage, subjects' performance varied as a function of speech type, passage difficulty, and recognition question type. Subjects listening to synthetic speech responded more accurately to word recognition questions than to proposition recognition questions. Listeners who heard natural speech, in contrast, demonstrated better proposition recognition performance. The results indicate that listeners who heard synthetic speech attended more closely to the acoustic-phonetic input than to the propositions of the passages. The results are discussed in terms of a limited capacity attentional mechanism. [Work supported by NSF IRI 86-17847.]

3:45

9SP11. Context effects in the perception of personal information in the speech signal. John Mullennix (Dept. of Psychol., Wayne State Univ., Detroit, MI 48202), Keith Johnson (UCLA, Los Angeles, CA 90024), and Meral Topcu (Wayne State Univ., Detroit, MI 48202)

The speech signal contains linguistic, personal, and social information. Many studies have demonstrated that the perception of linguistic information is subject to context effects. This paper is a report of a study concerning context effects in the perception of personal information. When listeners were asked to identify the speaker of synthetic stimuli (the vowel /i/) in terms of male/female attributes, their responses were most affected by F_0 and formant values with only a small effect of glottal waveform shape. The results of a perceptual anchoring study will be reported, in which listeners were asked again to identify the stimuli on the basis of speaker attributes, but with one endpoint of the synthetic continuum presented more often than any of the other stimuli. The results of this experiment will be discussed in terms of the hypothesis that listeners' perceptions of personal information in the speech signal are influenced by context. [Work supported by NIH.]

4:00

9SP12. On the perceptual differentiation of spontaneous and prepared speech. Robert E. Remez, Stefanie M. Berns, Jennifer S. Nutter, Jessica M. Lang, Lila Davachi (Dept. of Psychol., Barnard College, 3009 Broadway, New York, NY 10027), and Philip E. Rubin (Haskins Labs., 270 Crown St., New Haven, CT 06511)

Naive listeners are readily able to differentiate spontaneously produced speech from speech produced from text. The prior studies have employed lexically, syntactically, and thematically identical pairs of natural sentences extracted from brief fluent monologs (< 40 s in duration), finding relatively high levels of performance in tests of perceptual differentiation. To determine which attributes of the speech signal contribute to the perceptual differentiation of spontaneous and prepared

speech, the present study manipulated several likely acoustic parameters employing techniques of speech synthesis. One condition reduced the frequency variation of the synthetic copies of the utterances to a monotone. A second condition removed the segmental attributes (consonants and vowels) from the sentence pairs by low-pass filtering of the synthetic signals, leaving metrical and fundamental frequency variation intact. The final condition neutralized both the segmental and phonatory attributes, leaving only metrical properties available to perceivers by which to differentiate the sentence pairs. Although systematic perceptual effects were anticipated, in fact these acoustic conditions modulated the differentiability of individual sentence pairs in different ways. Evidence of this kind indicates that perceptual analysis of spontaneity takes place at the level of the sentence, and comparisons across the set of conditions prove that no single acoustic emblem of the speech signal conveys spontaneity to the listener.

4:15

9SP13. Further investigation of the semantic and pragmatic effects on speech production. Jan Charles-Luce and Basiliki Papadimitriou (Dept. of Commun. Disord. and Sci., SUNY, Buffalo, NY 14260)

Previous studies have shown that preceding semantic information affects the production of target words, for example blocking phonological rules or modifying duration. In addition, there is evidence that speakers modify their production of words during the speaker/listener exchange, for example in formal versus informal speaking situations. In the present study, two experimental environments were established to address the effects of semantic versus pragmatic contexts directly. In both environments, a subject was visually presented with a prime and then a target. The prime was either semantically related or unrelated to the target. The subject said the prime and target words aloud. In one experimental environment, a second person was present in the room to

which the subject was to communicate. In the other environment, a subject performed the task alone. Dependent variables were duration of, fundamental frequency of, and reaction time to onset of pronunciation of the target word. The results will be discussed in the frameworks of interactive activation and pragmatic compensation.

4:30

9SP14. Analysis of hesitations in spontaneous speech. D. O'Shaughnessy (INRS-Telecommunications, Nuns Island, Quebec H3E 1H6, Canada)

Spontaneous speech differs from read speech in several ways, especially in hesitation phenomena. This paper reports results on hesitation pauses (filled and unfilled) and restarts. For comparison purposes, the acoustic correlates of (unintended) hesitation pauses are compared to those for intentional pauses. A distinction is made between grammatical pauses (at major syntactic boundaries) and ungrammatical ones. Such pause types cannot be separated based on silence or prepausal duration, but rather in the pitch of the prepausal word. Ungrammatical pauses tended to have few F_0 continuation rises, whereas virtually all grammatical pauses were accompanied by a prior F_0 rise of at least 10 Hz. While silent pauses are easy to locate in speech recognition applications, filled pauses (e.g., "err," "umm") resemble words in continuous speech. Filled pauses at major syntactic boundaries were about 300–450 ms, whereas those within syntactic units were shorter. Filled pauses had falling or flat and low F_0 patterns. Ones at syntactic boundaries tended to start higher in F_0 and then fall, whereas filled pauses internal to a syntactic unit had lower F_0 patterns. Concerning restarts in spontaneous speech, when a word was completely repeated, it had virtually the same prosodics in both its instances. When a word was changed in the restart, its second instance was more stressed. [Work supported by Canadian government.]

FRIDAY AFTERNOON, 3 MAY 1991

LIBERTY B, 12:55 TO 4:45 P.M.

Session 9UW

Underwater Acoustics: Bubbles and Ambient Noise

Michael Longuet-Higgins, Chair
La Jolla Institute, P.O. Box 1434, La Jolla, California 92038

Chair's Introduction—12:55

Contributed Papers

1:00

9UW1. Propagation theory of extended noise sources: Improvements and new developments. C. David Rees (Ocean and Atmospheric Sci. Div., Code 541, Naval Ocean Syst. Ctr., San Diego, CA 92152-5000)

Both understanding the source generation characteristics of ocean surface noise and other extended sources from the viewpoint of oceanographic acoustics, and achieving a useful understanding of the charac-

teristics of a noise field at a sensor array to discriminate against noise, require accurate modeling of source to sensor propagation. Kuperman and Ingenito [J. Acoust. Soc. Am. 67, 1988–1996 (1980)] laid out a general Greens function technique for the totally range independent vertically stratified case, but treated in detail only a totally uncorrelated infinite sheet source. The author has developed and analyzed extensions to this approach, including specified source correlation functions and extended sources of finite extent. Development of a range-dependent propagation theory seems to be required to achieve the observed noise

peak at near-horizontal angles. The parabolic equation formalism provides a natural basis for this: techniques and results developed by the author to date are discussed and presented.

1:15

9UW2. Three-dimensional rough interface scattering of ambient noise. Jin-Yuan Liu and Henrik Schmidt (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

Based on the previously developed theory of perturbation approach to rough surface scattering [W. A. Kuperman and Henrik Schmidt, *J. Acoust. Soc. Am.* **86**, 1511–1522 (1989)] and a model for the noise field generated by surface random sources in an ocean waveguide [W. A. Kuperman and F. Ingenito, *J. Acoust. Soc. Am.* **67**, 1988–1996 (1980)], a formulation for rough interface scattering of surface-generated ambient noise in a horizontally stratified ocean is established. The result is then applied to study the three-dimensional scattering in a shallow-water waveguide environment bounded below by a viscoelastic medium. The three-dimensional spatial correlation of the reverberated noise in the waveguide and the scattered noise in the elastic medium are examined in terms of the relation to frequency, water depth, and rough interface statistics. It is demonstrated that even in deep water, the scattering of surface generated noise into interface waves (Scholte waves) is important, showing these waves carriers of ambient noise energy consistent with experimental observation. [Work supported by NOARL.]

1:30

9UW3. Trends of wind and noise dependency at Lake Pend Oreille, Idaho. Jonathan Cummings, II (David Taylor Res. Ctr., Bethesda, MD 20084)

Ambient noise and wind-speed measurements were obtained at Lake Pend Oreille periodically from October 1985 through August 1989. Cyclic trends in noise levels were observed and are summarized. Variations in noise level with time of day and season of the year correlate well with wind speeds and trends in recreational boating use. Daily averaged data show the contribution of recreational boating to lake noise: 10–15 dB average level increase for midday as compared to night. Average seasonal trends are generally higher (up to 10 dB) during the summer months as compared to winter months. Noise dependencies on wind speed were empirically derived and are found to compare well with observations from ocean environments.

1:45

9UW4. Sound radiation from large raindrops: Dependence on salinity and temperature. Peter W. Jacobus,^{a)} Herman Medwin, and Jeffrey A. Nystuen (Physics Dept., Naval Postgraduate School, Monterey, CA 93943)

Raindrops are known to produce bubbles by at least two different mechanisms: type I, the pinch-off of the bottom of the crater, and type II, air entrainment from a turbulent water jet formed as a part of the drop splash. Bubble energy is shown to dominate impact energy for the useful range of type II drops. Previous work [Snyder *et al.*, *J. Acoust. Soc. Am. Suppl.* **1** **88**, S2 (1990)] has shown a relationship between drop size and bubble frequency of the type II mechanism. The onset of this mechanism is related to the drop kinetic energy at impact, rather than to velocity. As the drop kinetic energy increases above a threshold of 2×10^{-4} J, the likelihood of bubble entrainment increases to approximately 65%. Further work reveals the effects of temperature, salinity, and surface tension on the sound radiated from large raindrops. The relation between the drop diameter and the spectrum of the underwater acoustic energy is examined for terminal velocity drops. This relation makes possible the remote measurement of both drop size distribution and rainfall rate. [Work supported by ONR.] ^{a)}LT, USN.

2:00

9UW5. Bubble production by capillary waves. Ali Kolaini, Ronald A. Roy, and Lawrence A. Crum (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

Bubble-producing capillary waves can be easily generated in a laboratory tank (8 cm × 22 cm × 150 cm) by blowing air over the water. The lowest wind speed required for the occurrence of this process is measured. The generation of the capillary waves depends solely on the surface tension, which can be changed by adding surfactants and other chemicals to the tank water. The effect of surface tension on the wind-speed threshold for bubble production is discussed. Using a coincidence detector, the bubble production rates per unit area can be measured. Lowering the surface tension, surprisingly, increases the rate of bubble production. The dependence of the bubble production rate on wind speed and wind fetch is also discussed. The underwater acoustic emissions from these bubbles are measured. Average power spectrum for several wind speeds are obtained that exhibit a broad range of radiated frequencies (i.e., bubble sizes) and a weak dependence on the wind speed. [Work supported by ONR, ONT, and AEAS.]

2:15

9UW6. The underwater sound produced by the impact of solid objects. Michael Nicholas, Ronald A. Roy, and Lawrence A. Crum (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

There have been many studies that indicate that there is significant underwater sound produced by precipitation [e.g., J. A. Scrimger, *Nature* **318**, 647 (1985)]. It has also been demonstrated that the major contribution to the underwater sound of rainfall is that due to the entrainment of gas bubbles by the impacting drop [Pumphrey *et al.*, *J. Acoust. Soc. Am.* **31**, 1080 (1989)]. In an attempt to understand the underwater sound produced by hail, the impact of solid objects with a plane water surface have been examined. With the aid of a high-speed movie camera and underwater transducers, it has been determined that the air entrained by the impacting object plays a major role in the sound production. The details of this process as well as the implications concerning the underwater sound produced by hail will be presented. [Work supported by ONR.]

2:30

9UW7. The hydrodynamic and acoustic behavior of bubble plumes generated by an impacting water jet. Ali Kolaini, Ronald A. Roy, and Lawrence A. Crum (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

The preliminary results of an experimental study of the underwater sound field emitted by a bubble plume generated by dropping fixed volume of water, held in a cylindrical container, onto a still water surface were previously presented [*J. Acoust. Soc. Am. Suppl.* **1** **88**, 514 (1990)]. Further studies of the acoustic and hydrodynamic characteristics of the bubble plume are presented. The high-speed video images reveal the formation of a cylindrical plume that grows in length until all of the impacting water volume is injected into the still water. As the leading end of the plume advances, a “substructure” separates from the rest of the plume. The onset of the large-amplitude, low-frequency sound emission occurs at the instant the substructure detaches. The resonance frequencies of the densely populated substructures are inversely proportional to their radii and are highly dependent on the void fraction. Experimental results are presented which show that detached plumes undergo damped volume oscillations. The measured damping coefficients are found to be constant and are believed to be related to the thermal damping coefficient. The dependence of the detached plume diameters and undetached plume lengths on the containers radii, lengths, and heights above the still water level are also discussed. [Work supported by ONR and ONT.]

3:00

9UW8. Some nonlinear aspects of the frequency response of a strongly excited gas bubble oscillator. Dong H. Kim (Acoust. and Vibration Lab., Korea Standards Res. Inst., P.O. Box 3, Daedok Sci. Town, Daejeon, Korea)

A numerical analysis is carried out for the nonlinear phenomena of the bubble oscillator. The model is based on the Keller's formulation for the bubble dynamics. Interpretation of the bubble interior is based on the formulation by Prosperetti. His formulation adopts the energy equation for the analysis of the bubble interior. The numerical simulation shows typical nonlinear phenomena in its frequency response. Among such nonlinear aspects are the jump phenomenon, the hysteresis effect, the shift of natural frequency of the system, and the appearance of superharmonic resonances. It is deduced that the nonlinear frequency response is dependent upon the initial condition of the bubble oscillator, and some multivalued frequency region can appear in the response curve. Nonlinear phenomena that appeared in the bubble oscillator are compared with those of the Duffing equation and it may be said that the bubble dynamic equation has similar nonlinear aspects to the Duffing equation.

3:15

9UW9. Multiple scattering from a randomly distributed cloud of bubbles. Lintao Wang and Kenneth E. Gilbert (Natl. Ctr. for Physical Acoust., University, MS 38677)

Exact multiple scattering calculations are presented for scattering from a randomly distributed cloud of bubbles. The conditions under which a single scattering (Born) approximation holds are discussed. In addition, numerical calculations are presented that show the relative contributions of coherent and incoherent scatter. The multiple scattering calculations are compared to scattering from an "effective fluid" model of a bubble cloud. It is shown that while the fluid model is adequate for predicting forward scatter, it cannot accurately predict backscatter when the acoustic wavelength is comparable to the dimensions of the bubble cloud. Some alternative approaches are discussed for accurately predicting backscatter from a randomly distributed cloud of bubbles. [Work supported by ONR.]

3:30

9UW10. Pulse length effects on the transmissivity of bubbly water. H. R. Suiter (Naval Coastal Syst. Ctr., Code 2120, Panama City, FL 32407)

The passage of sound through bubbly water is strongly attenuated by scattering and absorption. Such attenuation is most severe around the frequency of resonance of individual bubbles. A bubble takes a finite time to ring up to steady-state conditions and continues to oscillate for a finite time after the driving pressure ceases. Low backscatter for short pulse lengths has been observed in near-surface seawater [Akulichev *et al.*, Sov. Phys. Acoust. 32, no. 3]. An experiment is described that looked for a corresponding enhancement in transmission. Comparisons were made between the attenuations of brief waveform bursts and longer bursts. The frequency range of this experiment was 50-200 kHz. The bubbles were made by the electrolysis of fresh water in a small laboratory tank. For bursts of 6-20 wavelengths in duration, no difference in the attenuations was discerned in comparison with a 2.7 wavelength duration burst. [Work supported by ONR.]

3:45

9UW11. The release of air bubbles from an underwater nozzle. Michael Longuet-Higgins (La Jolla Inst., P.O. Box 1434, La Jolla, CA 92038 and Inst. for Nonlinear Sci., Univ. of California, San Diego R-002, La Jolla, CA 92093), Bryan R. Kerman (Canada Center for Inland Waters, Burlington, Ontario, Canada), and Knud Lunde (Dept. of Appl. Math. and Theoretical Physics, Cambridge CB3 9EW, England)

Air bubbles released from an underwater nozzle emit an acoustical pulse that is of interest both for the study of bubble detachment and for elucidating the mechanism of sound generation by a newly formed bubble. In this paper, the sequence of bubble shapes is calculated theoretically from a given nozzle, and it is shown that there is for each nozzle a bubble of maximum volume V_{\max} . Assuming that the bubble becomes detached at its "neck," and that the volume of the detached bubble equals the volume V^* of the undetached bubble above its "neck," it is determined for each nozzle diameter D an acoustic frequency f^* corresponding to "slow" bubble release. Experiments show that the acoustic frequency, hence the bubble size, depends on the rate of air flow to the bubble, but for slow rates of flow the frequency f is very close to the theoretical frequency f^* . High-speed photographs suggest that when the bubble pinches off, the limiting form of the surface is almost a cone. This is accounted for by assuming a line sink along the axis of symmetry. Immediately following pinch off, there is evidence of the formation of an axial jet going upward into the bubble. This may play a part in stimulating the emission of sound.

4:00

9UW12. Numerical simulations of bubble release from underwater needles. Hasan N. Oguz and Andrea Prosperetti (Dept. of Mech. Eng., Johns Hopkins Univ., Baltimore, MD 21218)

The release of bubbles from a submerged needle is a noisy process that is difficult to stabilize. Longer needles are found to produce a much steadier stream of bubbles than short ones. A possible explanation for this behavior is given by means of a dynamic model that accounts for the pressure drop inside the needle. The flow field created by the injected bubble is assumed to be irrotational and a boundary integral formulation is used to calculate the evolution of the bubble surface. Realistic bubble formation histories are obtained and the numerical results appear to be in good agreement with the available experimental measurements. The behavior of the bubble after detachment and the effect of the previously released bubble are also studied. [Work supported by ONR.]

4:15

9UW13. Dynamics of air bubbles entrapped by capillary waves. Hasan N. Oguz (Dept. of Mech. Eng., Johns Hopkins Univ., Baltimore, MD 21218) and Michael S. Longuet-Higgins (Ctr. for Studies of Nonlinear Dynamics, La Jolla Inst., La Jolla, CA 92038)

It is well known that entrapment of air bubbles near the sea surface contributes substantially to the underwater noise levels. The collapse of air pockets created by steep capillary waves is a possible mechanism of bubble formation. A boundary integral formulation is employed to simulate the bubble behavior after detachment from these air pockets. The initial profile of the trapped bubble, approximated from the form of the capillary waves, suggests a high elongated bubble relatively far from the

sea surface. A sequence of initial bubble shapes is generated from successively closer approximations to the bubble shape. Depending on the initial curvature of the detachment point the bubble may break up into two or oscillate as a single bubble in a rather violent manner. Volume oscillations that are responsible for the radiation of sound are found to be affected by the shape oscillations. It is found that a traveling capillary wave on the bubble surface can cause a pressure pulse upon reaching the axis. As a result, the pressure signal may deviate substantially from the simple damped sinusoid of a spherical bubble. [Work supported by ONR.]

4:30

9UW14. Direct numerical simulation of bubble behavior in oscillatory flow. D. Z. Zhang, A. Prosperetti (Dept. of Mech. Eng.,

Johns Hopkins Univ., Baltimore, MD 21218), and A. S. Sangani (Syracuse Univ., Syracuse, NY 13244)

The motion of spheres (bubbles) in an incompressible liquid undergoing a small-amplitude oscillatory motion is calculated by a multipole expansion method. In the limit of small viscosity, the Stokes layer is confined to the vicinity of the surface of the bubbles, which therefore interact approximately only through the pressure field. The motion of the spheres is parametrized in terms of added mass, Basset, and drag forces; the coefficients of which are obtained from the simulation. To obtain results useful for the study of pressure wave propagation in bubbly liquids, several bubble configurations are studied for different (finite) volume fractions and the results then averaged. The effects of surface tension and bubble density are also considered. [Work supported by DOE.]